

UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE PATENT TRIAL AND APPEAL BOARD

ZTE USA, INC.
Petitioner

v.

SAINT LAWRENCE COMMUNICATIONS LLC
Patent Owner

Case: IPR2016-00705

Patent 7,260,521

DECLARATION OF MICHAEL T. JOHNSON, PHD

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EXHIBITS CITED IN THIS DECLARATION

Description	Exhibit #
U.S. Patent No. 7,260,521	1001
File history of U.S. Patent No. 7,260,521	1002
ITU-T Recommendation G.729, "Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP)," March 1996	1004
ITU-T Recommendation G.728, "Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction," September 1992	1005
U.S. Patent No. 5,797,119 ("Ozawa")	1006
U.S. Patent No. 5,235,669 ("Ordentlich")	1007
U.S. Patent No. 5,444,816 ("Adoul")	1008
Atal, B. and Remde, J., "A New Model of LPC Extension for Producing Natural-Sounding Speech at Low Bit Rates," <i>IEEE</i> , pp. 614-617 (1982)	1009
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Schroeder, M. and Atal, B., "Code-Excited Linear Prediction (CELP): High Quality Speech at Very Low Bit Rates," <i>IEEE</i> , pp. 937-940 (1985)	1012
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Ramachandran, R. and Kabal, R., "Pitch Prediction Filters in Speech Coding," <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 37(4):467-478 (1989)	1019
Kleijn, W. and Ketchum, R., "Improved Speech Quality and Efficient	1020

Vector Quantization in SELP,” <i>IEEE</i> , pp. 155-158 (1988)	
Marques, J, et al., “Improved Pitch Predication With Fractional Delays in CELP Coding,” <i>Filters in Speech Coding</i> ,” <i>IEEE</i> , 665-668 (1990)	1021
Chen, J., and Gersho, A., “Adaptive Postfiltering for Quality Enhancement of Coded Speech,” <i>IEEE</i> , 3(1), pp. 59-71 (1995)	1022
ITU G.722 – General Aspects of Digital Transmission Systems Terminal Equipments (1988)	1023
Cheng, Y., et al., “Statistical Recovery of Wideband Speech From Narrowband Speech,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 2(4):544-548 (1994)	1024
Ordentlich, E. and Shoham, Y., “Low-Delay Code-Excited Linear-Predictive Coding of Wideband Speech at 32 Kbps,” <i>IEEE</i> , pp. 9-12 (1991)	1025
Avendano, C., et al., “Beyond Nyquist: Towards the Recovery of Broad-Bandwidth Speech from Narrow-Bandwidth Speech,” <i>Eurospeech</i> , 95 (1995)	1026
J. Schnitzler, A 13.0 KBIT/S Wideband Speech Codec Based on SB-ACELP, <i>IEEE</i> , pp. 157-161 (1998)	1027
U.S. Patent 5,293,449 (“Tzeng”)	1028
U.S. Patent 4,868,867 (“Davidson”)	1029
Joint Claim Construction and Prehearing Statement, <i>Saint Lawrence v. LG Electronics, Inc. – 14cv1055</i> (EDTX) dated July 17, 2015	1030
Kroon, P., “Regular-Pulse Excitation – A Novel Approach to Effective and Efficient Multipulse Coding of Speech,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 34(5):1054-1063 (1986)	1031
Atal, B. and Schroeder, M., “Predictive Coding of Speech Signals and Subjective Error Coding,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 27(3):247-254 (1979)	1032
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Oppenheim and Schaffer, <i>Discrete-Time Signal Processing</i> , pp. 17-33 (1989)	1035

I. INTRODUCTION

1. My name is Dr. Michael T. Johnson, and I have been retained by the law firm of Finnegan, Henderson, Farabow, Garrett & Dunner, LLP on behalf of ZTE USA, Inc. as an expert in the relevant art.

2. I have been asked to provide my opinions and views on the materials I have reviewed in this case related to Ex. 1001, U.S. Patent No. 7,260,521 (“the ’521 Patent”) (“the patent-at-issue”), and the scientific and technical knowledge regarding the same subject matter before and for a period following the date of the first application for the patent-at-issue was filed.

3. I am compensated at the rate of \$350/hour for my work, plus reimbursement for expenses. My compensation has not influenced any of my opinions in this matter and does not depend on the outcome of this proceeding or any issue in it.

4. My opinion and underlying reasoning for this opinion is set forth below.

A. Background And Qualifications

5. I am currently a Full Professor in the Department of Electrical and Computer Engineering at Marquette University in Milwaukee, Wisconsin. I received my PhD from Purdue University in West Lafayette, Indiana, in 2000, with a doctoral dissertation focused on speech processing and automatic speech recognition. Prior to receiving my PhD, I worked in industry as a design engineer and engineering manager for more than 5 years, primarily in the telecommunications industry.

6. The focus area of my research work is Speech and Signal Processing, and I regularly conduct and publish original research articles within this field, with a total publication record including more than 30 peer-reviewed journal articles and more than 90 total peer-reviewed publications over the past two decades. My specific expertise includes speech signal processing and enhancement, automatic speech recognition, machine learning and pattern recognition, and bioacoustics. Current professional activities within this field include roles as a member of the IEEE Signal Processing Society Speech and Language Technical Committee and as an Area Editor for the journal Speech Communication in the area of speech enhancement.

7. A detailed record of my professional qualifications, including a list of publications, awards, and professional activities, is set forth in my curriculum vitae, attached to this report as Appendix A.

B. Information Considered

8. In addition to my general knowledge gained as a result of my education and experience in this field, I have reviewed and considered, among other things, the '521 Patent, the prosecution history of the '521 Patent, and the prior art of record.

9. The full list of information that I have considered in forming my opinions for this report is set forth throughout the report and listed in the attached Appendix B.

II. LEGAL STANDARDS

10. In forming my opinions and considering the patentability of the claims of the '521 Patent, I am relying upon certain legal principles that counsel has explained to me.

11. I understand that for an invention claimed in a patent to be found patentable, it must be, among other things, new and not obvious in light of what came before it. Patents and publications which predated the invention are generally referred to as "prior art."

12. I understand that in this proceeding the burden is on the party asserting unpatentability to prove it by a preponderance of the evidence. I understand that "a preponderance of the evidence" is evidence sufficient to show that a fact is more likely than not.

13. I understand that in this proceeding, the claims must be given their broadest reasonable interpretation consistent with the specification. The claims after being construed in this manner are then to be compared to information that was disclosed in the prior art.

A. Person Of Ordinary Skill In The Art

14. I have been informed that the claims of a patent are judged from the perspective of a hypothetical construct involving "a person of ordinary skill in the art." The "art" is the field of technology to which the patent is related. I

understand that the purpose of using a person of ordinary skill in the art's viewpoint is objectivity. Thus, I understand that the question of validity is viewed from the perspective of a person of ordinary skill in the art, and not from the perspective of (a) the inventor, (b) a layperson, or (c) a person of extraordinary skill in the art. I have been informed that the claims of the patent-at-issue are interpreted as a person of ordinary skill in the art would have understood them in the relevant time period (*i.e.*, when the patent application was filed or earliest effective filing date).

15. It is my opinion that a person of ordinary skill in the art relevant to the '521 patent would have familiarity and design experience with the most common types of speech encoders and decoders at the time of this patent, one of the most common of which was the Code Excited Linear Prediction approach to speech coding that is the topic of this declaration. That includes basic knowledge of linear prediction analysis, various types of fixed excitation codebook methods, long-term pitch prediction and adaptive codebooks, perceptual weighting, the optimization principles underlying analysis-by-synthesis speech coding, and bandwidth expansion. Such background knowledge would typically represent the equivalent of a Master's degree level of engineering in an engineering or computer science discipline combined with 3-5 years of industrial application experience.

16. I understand that a “person of ordinary skill is also a person of ordinary creativity, not an automaton” and have applied that standard in making judgments about persons working in speech modeling and coding.

B. Anticipation

17. I understand that the following standards govern the determination of whether a patent claim is “anticipated” by the prior art. I have applied these standards in my analysis of whether claims of the ’521 Patent were anticipated at the time of the invention.

18. I understand that a patent claim is “anticipated” by a single prior art reference if that reference discloses each element of the claim in a single embodiment. A prior art reference may anticipate a claim inherently if an element is not expressly stated, but only if the prior art necessarily includes the claim limitations.

19. I understand that the test for anticipation is performed in two steps. First, the claims must be interpreted to determine their meaning. Second, a prior art reference is analyzed to determine whether every claim element, as interpreted in the first step, is present in the reference. If all the elements of a patent claim are present in the prior art reference, then that claim is anticipated and is invalid.

20. I understand that it is acceptable to examine extrinsic evidence outside the prior art reference in determining whether a feature, while not expressly discussed in the reference, is necessarily present within that reference.

C. Obviousness

21. I understand that a claim can be invalid in view of prior art if the differences between the subject matter claimed and the prior art are such that the claimed subject matter as a whole would have been “obvious” at the time the invention was made to a person having ordinary skill in the art.

22. I understand that the obviousness standard is defined at 35 U.S.C. § 103(a). I understand that a claim is obvious over a prior art reference if that reference, combined with the knowledge of one skilled in the art or other prior art references discloses each and every element of the recited claim.

23. I also understand that the relevant inquiry into obviousness requires consideration of four factors:

- a. The scope and content of the prior art;
- b. The differences between the prior art and the claims at issue;
- c. The knowledge of a person of ordinary skill in the pertinent art; and
- d. Objective factors indicating obviousness or non-obviousness may be present in any particular case, such factors including commercial success of products covered by the patent claims; a long-felt need for the invention; failed

attempts by others to make the invention; copying of the invention by others in the field; unexpected results achieved by the invention; praise of the invention by the infringer or others in the field; the taking of licenses under the patent by others; expressions of surprise by experts and those skilled in the art at the making of the invention; and that the patentee proceeded contrary to the accepted wisdom of the prior art.

24. I understand that when combining two or more references, one should consider whether a teaching, suggestion, or motivation to combine the references exists so as to avoid impermissible hindsight. I have been informed that the application of the teaching, suggestion or motivation test should not be rigidly applied, but rather is an expansive and flexible test. For example, I have been informed that the common sense of a person of ordinary skill in the art can serve as motivation for combining references.

25. I understand that the content of a patent or other printed publication (i.e., a reference) should be interpreted the way a person of ordinary skill in the art would have interpreted the reference as of the effective filing date of the patent application for the '521 Patent. I have assumed that the person of ordinary skill is a hypothetical person who is presumed to be aware of all the pertinent information that qualifies as prior art. In addition, the person of ordinary skill in the art makes

inferences and creative steps. He or she is not an automaton, but has ordinary creativity.

26. I have been informed that the application that issued as the '521 patent claims priority to a foreign application that was filed on October 27, 1998. As a result, I will assume the relevant time period for determining what one of ordinary skill in the art knew is October 27, 1998, the effective filing date for purposes of this proceeding.

D. Claim Construction

27. I have been informed that a claim subject to *inter partes* review is given its “broadest reasonable construction in light of the specification of the patent in which it appears.” I have been informed that this means that the words of the claim are given their plain meaning from the perspective of one of ordinary skill in the art unless that meaning is inconsistent with the specification. I understand that the “plain meaning” of a term means the ordinary and customary meaning given to the term by those of ordinary skill in the art at the time of the invention and that the ordinary and customary meaning of a term may be evidenced by a variety of sources, including the words of the claims, the specification, drawings, and prior art.

28. I understand that in construing claims “[a]ll words in a claim must be considered in judging the patentability of that claim against the prior art.” (MPEP § 2143.03, citing *In re Wilson*, 424 F.2d 1382, 1385 (CCPA 1970)).

29. I understand that extrinsic evidence may be consulted for the meaning of a claim term as long as it is not used to contradict claim meaning that is unambiguous in light of the intrinsic evidence. *Phillips v. AWH Corp.*, 415 F.3d 1303, 1324 (Fed. Cir. 2005) (citing *Vitronics Corp. v. Conceptronic, Inc.*, 90 F.3d 1576, 1583-84 (Fed. Cir. 1996)). I also understand that in construing claim terms, the general meanings gleaned from reference sources must always be compared against the use of the terms in context, and the intrinsic record must always be consulted to identify which of the different possible dictionary meanings is most consistent with the use of the words by the inventor. *See, e.g., Ferguson Beauregard/Logic Controls v. Mega Systems*, 350 F.3d 1327, 1338 (Fed. Cir. 2003) (citing *Brookhill-Wilk 1, LLC v. Intuitive Surgical, Inc.*, 334 F.3d 1294, 1300 (Fed. Cir. 2003)).

III. TECHNOLOGY BACKGROUND

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

30. The primary technology on which the '521 Patent is based is the field of speech coding. The principles of digital speech coding go back to the mid-1970's, having been established in the early days of the computer revolution (Ex. 1009,

Atal 1982). There are a number of comprehensive references from that time period covering the basics of speech coding, including (Ex. 1010, Spanias 1994).

31. Nearly all speech coding technology is still based on the same underlying principles, in which a speech sound pressure signal recorded by a microphone is analyzed and split into an excitation source component and a vocal tract filter component. As shown in Figure 1 below, this is based on the mechanism of human speech production: the excitation represents the periodic vibration of the vocal folds driving voiced sounds such as vowels or the constrictive airflow driving unvoiced sounds such as the consonants ‘f’ or ‘s’, while the vocal tract filter represents the frequency filtering/shaping of the articulators such as the tongue and lips, jointly referred to as the “vocal tract”, which together give individual speech sounds their fundamental time and frequency characteristics.

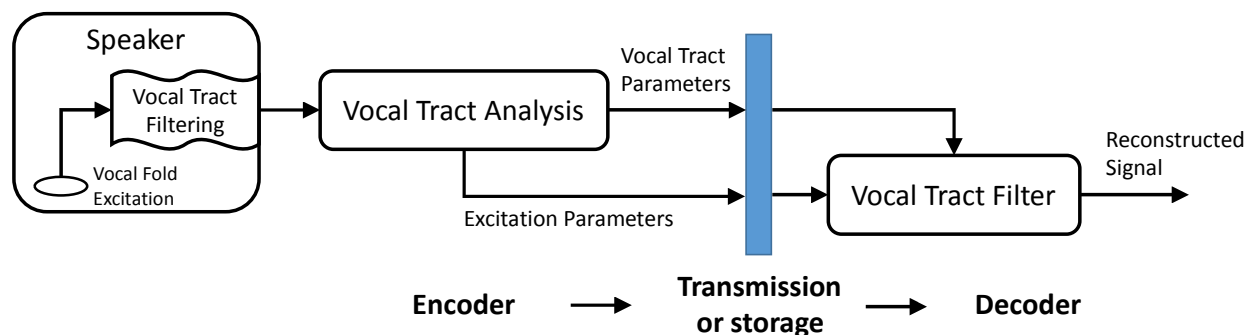


Figure 1 – Source filter model for speech coding and decoding.

32. As shown above, the basic principle of speech coding is to separate the source and filter components at the encoder and then compress these separately using as few bits as possible for efficient transmission or storage (Ex. 1010,

Spanias 1994). At a decoder, the speech signal can be re-synthesized by first reconstructing the excitation component and then passing the excitation through the vocal tract filter.

33. As described in Spanias's 1994 overview paper, the first stage of speech coding is to divide the signal into individual sections, called windows or frames, to encapsulate a small part of the signal containing approximately one single speech component (Ex. 1009, Atal 1982; Ex. 1010, Spanias 1994). Typically some initial pre-processing is applied, such as subtracting the mean and applying a simple filter, called a pre-emphasis filter, for focusing on the important frequency characteristics (Ex. 1010, Spanias 1994). These pre-emphasis filter or other pre-processing steps can be easily reversed at the decoder when the signal frames are connected back together for final output.

34. Because filter-related components change more slowly than the excitation related components, it is more efficient to transmit the associated parameters at two different rates. To implement this, the signal is divided first into longer frames and then further divided into shorter frames, sometimes called subframes, as implemented in some of the earliest speech coding standards such as the 1991 Federal Coding Standard 1016 (Ex. 1010, Spanias 1994; Ex. 1011, Federal Coding Standard 1016 1991). Except for the initial LPC analysis described in more detail

below, all processing occurs at the shorter frame level, and for simplicity in this document these subframes are referred to simply as “frames”.

35. Once pre-processing is complete, the signal is analyzed using Linear Prediction Coding (LPC) analysis, first applied to speech processing in the late 1960’s and early 1970’s (Ex. 1010, Spanias 1994), which models the overall vocal tract frequency characteristics of the sound and represents these using a compact set of parameters called LPC coefficients. LPC analysis is used because it is both effective and efficient to implement using the Levinson-Durbin algorithm. Once the LPC coefficients have been calculated, they may be directly quantized and transmitted, typically once for each long frame of the speech signal. Many current and historical speech coding methods use LPC analysis and LPC or equivalent coefficients to represent the vocal tract characteristic parameters for transmission or storage (Ex. 1010, Spanias 1994).

36. Once LPC coefficients are computed, each speech signal frame is filtered to remove the vocal tract characteristics, resulting in a residual signal that represents the original excitation component. The filter necessary to remove vocal tract characteristics is equivalent to the inverse of the LPC filter, i.e. its reciprocal in terms of frequency characteristics (Ex. 1009, Atal 1982). The impulse response of the necessary filter is computed directly from the LPC coefficients. Additionally, because the filter response extends across frame boundaries, a portion of the filter

output needs to come from the previous frame, and this is implemented by saving the previous frame output in memory, in the form of the “zero-input impulse response” and combining it with the impulse response of the current frame as described in the 1992 G.728 speech coding standard (Ex. 1005, G.728 1992). The LPC analysis, impulse response, and zero-input impulse response combine to implement LPC-based filtering, and output a signal representing excitation of the speech waveform (e.g. ITU G.728 1992), which remains to be encoded.

37. However, this remaining excitation component is difficult to represent compactly, and has significant impact on the quality of re-synthesized speech. As a result much of the core research in the area of speech coding since the mid 1970’s has related to finding a good excitation representation (Ex. 1010, Spanias 1994), one of the most common of which is CELP coding, to be described in the next section.

B. Code Excited Linear Prediction (CELP)

38. One of the first successful approaches for compactly representing speech excitation is Code Excited Linear Prediction (CELP), which is generally attributed to the research work of Bishnu Atal first published in 1985 (Ex. 1012, Schroeder 1985). CELP has been a widely used approach for speech coding approach, and it continues to be used in a wide variety of digital transmission and communication technologies (Ex. 1010, Spanias 1994; Ex. 1013, Hokanen 1997; Ex. 1014, G

06.60 1996; Ex. 1023, ITU G.722.2 2002; Ex.1015, ITU G.728 1992; Ex. 1004, ITU G.729 1996).

39. Atal's 1985 CELP method (Ex.1012, Schroeder 1985), shown in Figure 2 below, incorporated two basic techniques for encoding excitation effectively: 1) the use of a vector quantization codebook to select stored excitation waveforms, which can be transmitted using a compact codeword (Ex. 1012, Schroeder 1985), and 2) the use of an analysis-by-synthesis approach in the encoder to select the best codeword (Ex. 1031, Kroon 1986).

40. In the vector quantization approach described by Gersho and Cuperman in 1983, a number of different excitation signal vectors are stored in a simple pre-determined list, called a codebook, that identifies them by an identifying entry number, called a codeword. Rather than transmitting an entire signal, only the codeword needs to be transmitted, and can then be reconstructed at the decoder using a simple codebook look-up. Typically the number of codebook entries is a power of 2, such as 256 (8 bits), 512 (9 bits), or 1024 (10 bits), so that all possible codeword entries for a specific codeword length are utilized. To encode the signal, this codebook, called the excitation codebook, the innovation codebook, or sometimes simply the fixed codebook, is searched to identify the best codeword. The process of creating the codebook is done through a training process ahead of

time, using many examples of typical speech across a wide variety of speakers and speaking styles to identify the best set of codewords to use.

41. The analysis-by-synthesis idea, first characterized by Kroon in the late 1980's (Ex. 1031, Kroon 1986) is shown in Figure 2 below, essentially implements this search by building an extra decoder inside the encoder itself. The encoder implements an iterative loop, trying different excitation codewords or other coding parameters and using its internal decoder to measure the quality of the resulting signal, which it can easily do since it also has the original waveform available. The best codewords and parameters are then selected to represent the signal, and transmitted. The analysis-by-synthesis concept can be applied to any kind of signal encoding parameters, allowing the encoder to “try out” parameters and measure how they work before selecting them.

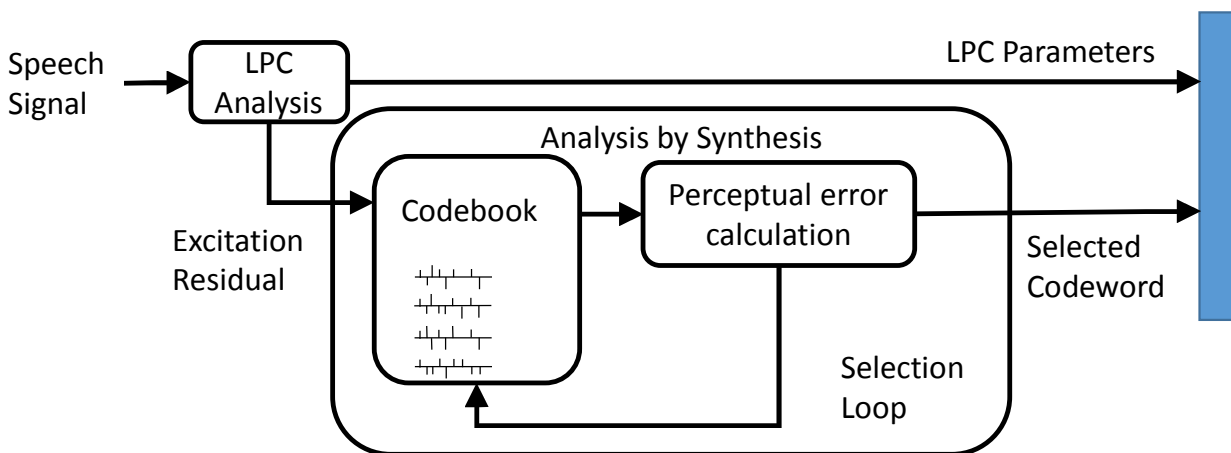


Figure 2 Basic CELP encoder

42. CELP coding typically incorporates an error measure that is based on a model of human speech quality perception, through the use of a perceptual

weighting filter in the error criteria that pre-dates Atal's original 1985 CELP paper (Ex. 1012, Schroeder 1985). In addition, there many variations on the basic CELP coding approach, most of which involve the implementation details of one or more vector quantization codebooks (Ex. 1027, Chan 1992) or additional signal processing related to the pitch of the speech signal, specifically the subtraction of a long-term prediction excitation component based on pitch prediction using an adaptive codebook, as shown by Singhai and Atal in 1984 (Ex. 1018, Singhai 1984; Ex. 1019, Ramachandran 1989).

C. Perceptual Weighting

43. Given the analysis-by-synthesis approach, the goal of the encoder is to find parameters that will most accurately reconstruct the original signal. However, what is most important is not the exact error, but rather how accurately listeners perceive the reconstructed signal. One of the earliest and simplest ways to represent listener perception into speech coding was by incorporating a simple perceptual weighting filter (Ex. 1032, Atal 1979) that emphasizes the frequency components that are most important to speech, prior to calculating error in the analysis-by-synthesis module. Such a filter was incorporated in Atal's original CELP idea (Ex. 1012, Schroeder 1985). This can be implemented in several ways since the perceptual weighting filter can be placed in more than one location within the signal path (e.g.

Ex. 1012, Schroeder 1985; Ex. 1010, Spanias 1994; Ex. 1011, Federal Standard 1016), but the underlying principle and result is the same.

D. Long term pitch prediction using adaptive codebook

44. The excitation component to be represented by the innovation codebook actually includes several distinct types of excitation components, which can be thought of as the “voiced”, or pitch-related, and “unvoiced” contributions to the signal. As shown by Singhai and Atal in 1984, removing the periodic part of the signal corresponding to the pitch prior to using the innovation codebook can substantially reduce error (Ex. 1018, Singhai 1986; Ex. 1019, Ramachandran 1989). To do this, it is necessary to implement a pitch search module to estimate the pitch period T and gain b (related to pitch strength), and use these to create a long-term prediction signal that can be subtracted to create a smaller excitation signal for the fixed codebook stage. The long-term prediction filter is typically a one-coefficient, or one-tap, filter with a transfer function of $1/(1-bz^{-T})$ (Ramachadran 1989), where the b parameter can be thought of as the pitch gain. Described in detail by Ramachadran in 1989, it is also possible to use a slightly more descriptive three-tap filter that includes additional coefficients near the pitch period (Ex. 1019, Ramachadran 1989), but this then requires that parameters be transmitted representing the additional coefficients.

45. The necessary pitch parameters are typically determined using an initial analytical pitch estimate, for example using the well-known autocorrelation method (Ex. 1033, Rabiner 1977), followed by a search around nearby values using a squared error criteria. The process of obtaining the initial estimate is sometimes called an “open-loop search” because it is analytical rather than iterative, while the final selection is called a “closed-loop search” because it loops to find the best value (Ex. 1010, Spanias 1994)

46. There is a difficulty with direct implementation of the closed-loop pitch search for cases where the pitch period T is smaller than the current frame size. In this case the search process requires samples of the past excitation that extend into the current frame, which are not yet available. Kleijn’s 1988 paper on Stochastic Excited Linear Prediction (SELP) addressed this specific issue, replicating the excitation signal from the previous frame as an estimate for the current frame (Ex. 1020, Kleijn 1988). One way to view this process is as selection from an “adaptive codebook” component representing the past excitation signal, the term adaptive meaning that the codebook changes over time (Ex. 1020, Kleijn 1988; Ex. 1010, Spanias 1994).

47. Although the term “adaptive codebook” is historically used for this technique, as first described by Kleijn’s 1988 paper, it is somewhat of a misnomer and can lead to confusion, as there is no literal codebook and no codeword being

selected or transmitted (Ex. 1020, Kleijn 1988; Ex. 1010, Spanias 1994). The parameters for the long-term prediction component remain the pitch period T and pitch gain b .

48. Additionally, the pitch period T can be either an integer number of samples, or can be interpolated to be a fractional period pitch estimate which gives additional accuracy. Introduced by Marques in 1990 and later integrated into the G.729 coding standard (Ex. 1021, Marques 1990; Ex.1004, G.729 1996), fractional pitch estimation requires some additional signal processing at both the encoder and the decoder using an oversampling technique to construct the long-term prediction component, which includes oversampling of the past excitation signal in the adaptive codebook stage. Conceptually however, fractional pitch simply allows for an increased pitch resolution and therefore more accurate long-term prediction signal.

49. Regardless of exact method, the pitch period and pitch gain parameters are selected to minimize the squared error between the input excitation signal and the long-term prediction signal. The long-term prediction signal is subtracted from the input excitation signal and passed to the innovation codebook, which then uses the analysis-by-synthesis criteria to search for the best innovation codeword, as shown in Figure 3 below.

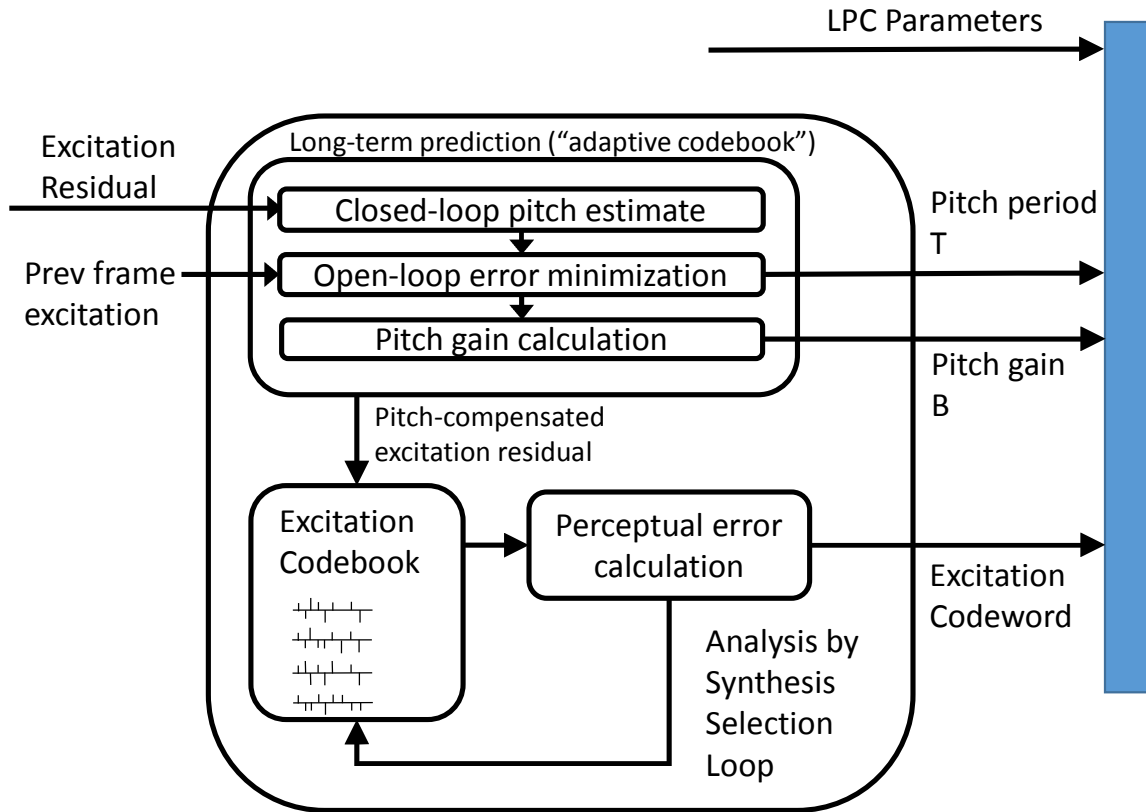


Figure 2 Illustration of long-term prediction coupled with analysis-by-synthesis loop

E. CELP Decoding

50. At the decoder, the CELP signal parameters can be directly used to reconstruct each individual signal frame. The fixed excitation codeword is used as an index into the fixed codebook to find the excitation waveform, which is filtered using the long-term prediction filter to add the pitch contribution. This final excitation signal is then filtered using the LPC coefficients (which are updated once for each longer full frame) to add vocal tract characteristics and re-create a close approximation to the original signal frame. As summarized in Spanias's 1994 overview paper, the individual frames are then filtered to undo the initial pre-

processing (e.g. pre-emphasis) and connected together for final continuous speech output (Ex. 1010, Spanias 1994).

51. There are additional techniques used in some decoders to try and further improve the overall signal quality, especially when the transmission rate and therefore number of bits is low. This can include the use of post-filters (Chen 1995), and also bandwidth extension techniques, described further below.

F. Bandwidth extension

52. Most of the important content for speech signals is represented in the lower frequency range of 300-3000Hz. Historically, the telephone network encoded audio using quantization and transmission of the telephone signal, sampled at 8 kbits/second, with no specialized speech coding, and implemented a corresponding filter to limit audio content to this range. Because of this, it was natural during the early days of speech coding to use only this frequency range, as it was considered to be sufficient. Thus most early speech coding systems encoded just this “narrowband” low frequency portion of the speech signal.

53. As additional frequency bandwidth became less expensive and more accessible, it was recognized that this limitation unduly limited both the quality and intelligibility of speech signals, and the frequency range used for coding speech increased. Several standards for CELP coding include specific support across a variety of frequency ranges and corresponding coding rates, most notably

the 1988 wideband CELP standard G.722 (Ex. 1023, ITU G.722 1988). The term “wideband” is itself general and may refer to a variety of frequency ranges, typically outside 300-3000Hz. One common usage is to refer to speech signals sampled at 16kHz (as opposed to the standard 8kHz rate) as wideband. This corresponds to a frequency content up to an absolute maximum of 8000Hz as shown by Cheng’s 1994 paper (Ex. 1024, Cheng 1994), and results in a more typical added wideband frequency range on the order of 3400-7000Hz (Ex. 1025, Ordentlich 1991).

54. Extending CELP to wideband signals requires no fundamental modifications of the coding structure, but rather simply requires re-evaluation and establishment of appropriate bit allocations for the new range, as illustrated in the 1988 G.722 wideband coding standard (Ex. 1025, Ordentlich 1991; Ex. 1023, ITU G.722 1988).

55. Without actually encoding the wideband signal directly, it is also possible to infer the general spectral shape of the higher frequency content of speech signals from that of the lower frequency content. This allows for some of the missing high frequency content of the speech signals to be added back in at the decoder, even though it was not represented by the transmitted information, in order to create a higher quality sound. This process is generally referred to as bandwidth extension as described by Cheng in 1994 and Avendana in 1995 (Ex. 1024, Cheng 1994; Ex.

1026 Avendano 1995) but is sometimes incorporated within the context of wideband coding as shown in Schnitzler's 1998 paper on wideband CELP (Ex. 1027, Schnitzler 1998). Regardless of context, such frequency extension can be accomplished in a simple manner by simply using the harmonic structure represented by the encoded LPC analysis (Ex. 1024, Cheng 1994; Ex. 1026, Avendano 1995; Ex. 1027, Schnitzler 1998), or more recently through a variety of more complex approaches that try to estimate the most likely spectral content of the higher frequency components. To implement, the original narrowband signal is simply resampled to match the desired sampling rate (Ex. 1026, Avendano 1995; Ex. 1027, Schnitzler 1998) and the new extended bandwidth signal is directly added to the original.

IV. THE '521 PATENT

A. Background Of The '521 Patent

56. The '521 Patent generally relates to the efficient digitally encoding of a wideband speech signal. The '521 Patent specifically relates to an improved pitch analysis device and method for long-term pitch prediction. To accomplish this, the '521 Patent discloses a low-pass shaping filter, selected from one of K predetermined frequency shaping filters, which is applied to the output of the pitch codebook. These filters shape the pitch harmonics for voiced signals to give better overall spectral quality across the full band. The best filter (or alternatively none at

all, if that is the best) is identified according the resulting minimum prediction error and selected. Bits representing the selected filter are transmitted to the decoder along with the other coding information, including the corresponding pitch codevector and gain.

57. The '521 Patent also relates to the application of this improved pitch analysis device in the context of a cellular communication system, a cellular mobile transmitter/receiver unit, a cellular network element, and a bidirectional wireless communication sub-system.

B. Prosecution History (Ex. 1002)

58. According to the face of the '521 Patent, U.S. Patent Application No. 09/830,114 ("the '114 application"), filed April 24, 2001, is the application which issued as the '521 Patent. The '521 Patent claims priority to foreign application filed on October 27, 1998. Ex. 1001, the '521 Patent, at face of patent.

59. In the first Office Action, on July 1, 2005, the Examiner rejected claims 1, 3, 7, 10, 12, 16, 19, 21, and 25 under §102(b) as anticipated by Lin (US 5,621,852). The examiner rejected claims 2, 4, 9, 11, 13, 18, 20, 22, 27, 28-31, 34, 36, 37-40, 43, 45, 46-49, 52, 54, 55-58, 61, and 63 under 103(a) as unpatentable over Lin in view of Abdoul (US 5,754,976). The examiner objected to claims 5-6, 8, 14-15, 17, 23-24, 26, 32-33, 35, 41-42, 44, 50-51, 53, 59-60, and 62 as depending from a

rejected base claim but would be allowable if rewritten in independent form including all the limitations of the base claim and intervening claims.

60. In a response on November 1, 2005, the applicant amended the claims. For example, the applicant amended “an optimal set of pitch codebook parameters” to “a set of pitch codebook parameters” in claims 1 and 10 and included “signal” before “path(s)” in claims 1-4, 10-13, 19-22, 29-31, 38-40, 47, 49, and 56-58, changed “large geographical area” in claims 28 and 55 to remove “large,” amended claim 46 to remove “a received including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal,” and removed “cellular” from “cellular network” in claims 47-54.

61. In response to the rejections, the applicant argued in part that “Lin’s binary codebooks 1 and 3 are not used to produce a set of pitch codebook parameters but are used to search an optimum innovation sequence. The optimum innovation sequence is identified by indices I and j of the codebooks 1 and 2. Accordingly, the binary codebooks of Lin do not form a pitch analysis device nor do they form signal paths that are associated to respective sets of pitch codebook parameters.” (Resp., at 32).

62. In an Office Action on June 8, 2006, the examiner rejected claims 1-63 under §101 as being directed to non-statutory subject matter because they merely

manipulate an abstract idea: “the claimed process, a series of steps to be performed by a computer, amounts to a manipulation of an abstract idea since the process fails to provide any pre- or post- computer process activity.” (Office Action, at 2). The examiner noted that claims directed to speech or audio signal processing would be consisted statutory subject matter.

63. In an interview summary on September 6, 2006, it was noted that the proposed amendments to independent claims 1 and 10 appears to address the 101 rejection.

64. In a response on November 8, 2006, to the June 8, 2006 Office Action, the applicant amended the claims. For example, the applicant added the following limitation to claim 1 “a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized” and made the following change “. . . sets of pitch codebook parameters representative of said digitized input audio data. ” The applicant made similar amendments to claim 10, reflecting the results of the discussions during the examiner interview.

65. A notice of allowance was mailed on February 8, 2007.

C. Asserted Claims

66. My analysis will focus on claims 1, 2, 5-8, 10, 11, 14, 15, 17, 28, 29, 32, 33, 35, 37, 38, 41, 42, 44, 55, 56, 59, 60, 62 of the '521 Patent. Of these, claims {1, 2, 5-8} represent the first set of claims related to the pitch search device. Some or all of these claims are then repeated in claims {10, 11, 14, 15, 17} within the context of a encoder, in claims {28, 29, 32, 33, 35} within the context of a cellular mobile transmitter/receiver unit, in claims {37, 38, 41, 42, 44} within the context of a network element, and in claims {55, 56, 59, 60, 62} within the context of a pitch analysis method.

67. Summarizing the first set of claims noted above, claim 1 is an overview description of the pitch analysis device, identified as a pitch search device containing at least two signal paths plus a selector for those paths. Claim 2 further specifies that at least one of the two signal paths not include a filter.

68. Claim 5 specifies that the pitch prediction error calculation of claim 1 include 4 subcomponents: a) a convolution unit, b) a pitch gain calculator, c) an amplifier, and d) a combiner circuit.

69. Claim 6 narrows claim 5 such that the pitch gain calculator uses a specific gain formula.

70. Claim 7 narrows claim 5 to specify that the filters be identified by a filter index, the codevectors be identified by a codevector index, and that the filter index, codevector index, and gains together form the pitch codebook parameters.

71. Finally, claim 8 narrows claim 1 to specify that the pitch prediction error energy is calculated and used to select the corresponding best signal path.

72. The specifics of the individual claims will be discussed in detail in the Prior Art Analysis section.

D. Claim Construction

73. I have been informed that in my analysis, I should interpret the claims to have their broadest reasonable construction in light of the specification.

74. The broadest reasonable construction of “pitch codebook” is a storage containing a sequence of previous sample values. *See* Ex. 1001, 10:24-38 (“... the pitch contribution can be seen as a pitch codebook containing the past excitation signal .”); 10:24-27 (“In this case, the pitch contribution to the excitation signal $u(n)$ is given by $bu(n-T)$, where the total excitation is given by $u(n)=bu(n-T)+gc_k(n)$ ”); 12:1-6 (“The pitch codebook search module **301** is responsive to the target vector x , to the open-loop pitch lag T_{OL} and to the past excitation signal $u(n)$, $n<0$, from memory module **303** to conduct a pitch codebook (pitch codebook) search minimizing the above-defined search criterion C .”).

75. The broadest reasonable construction of “pitch code vector” is a vector containing previous sample values selected from the pitch codebook. See Ex. 1001, 10:38-46 (“a pitch codebook vector $v_T(n)$ at pitch lag T . . . is built by repeating the available samples from the past excitation until the vector is completed); 12:7-9 (“From the result of the search conducted in module **301**, module **302** generates the optimum pitch codebook vector V_T ”); and 10:50-53 (“In this case, the vector $v_T(n)$ usually corresponds to an interpolated version of the past excitation, with pitch lag T being a non-integer delay”).

76. The broadest reasonable construction of “means for calculating a pitch gain” is a circuit configured to calculate the pitch gain using the equation set forth at 12:40-43. See, e.g., Ex. 1001, 2:24-29 (“fixed point implementation”); 2:31-37; 12:39-44 (“Each gain $b^{(i)}$ is calculated in a corresponding gain calculator **306** ⁽ⁱ⁾ in association with the frequency shaping filter at index j , using the following relationship: $b^{(i)} = x^t y^{(i)} / \|y^{(i)}\|^2$ ”); 7:40-43; 7:44-49; 9:34-38. Thus, the claimed structure is such a circuit. The claimed function of the “means for calculating” is “to calculate the pitch gain using the equation set forth at 12:40-43.” Ex. 1001, 2:31-37; 12:39-44.

77. The broadest reasonable construction of “means for calculating an energy of the corresponding pitch prediction error” is a circuit configured to use a multiplier and a subtractor to calculate an energy of the corresponding pitch prediction error.

See, e.g., Ex. 1001, 2:24-29 (“fixed point implementation”); 2:31-37; 12:35-39 (“To calculate the mean squared pitch prediction error $e^{(i)}$ for each value of $y^{(i)}$, the value $y^{(i)}$ is multiplied by the gain b by means of a corresponding amplifier **307**⁽ⁱ⁾ and the value $b^{(i)}y^{(i)}$ is subtracted from the target vector x by means of subtractors **308**⁽ⁱ⁾”); 7:40-43; 7:44-49; 9:34-38. Thus, the claimed structure is a circuit. The claimed function of the “means for calculating” is “to use a multiplier and a subtractor to calculate an energy of the corresponding pitch prediction error.” Ex. 1001, 2:31-37; 12:35-39.

78. The broadest reasonable construction of “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” is a circuit configured to use a comparator to comparing different values of the energy of the pitch prediction error. *See, e.g.,* Ex. 1001, 2:24-29 (“fixed point implementation”); 2:31-37; 12:45-47 (“In selector **309**, the parameters b , T , and j are chosen based on v_T or $v_f^{(i)}$ which minimizes the mean squared pitch prediction error e .”); 7:40-43; 7:44-49; 9:34-38.

79. The broadest reasonable construction of “wideband input signal” is an input signal having a bandwidth that extends at least up to 7000 Hz. *See* Ex. 1001, 1:26-34.

80. The broadest reasonable construction of “encoded wideband signal” is a wideband signal (as constructed above) that has been encoded. *See, e.g.*, Ex. 1001, 9:34-38.

V. PRIOR ART ANALYSIS

81. I now turn to the references applied in the grounds for rejections discussed in Petition for *inter partes* review. In my analysis, I will specifically address the following references.

82. A claim chart demonstrating where each limitation is disclosed in the prior art is attached to my declaration as Appendix C.

No.	Reference	Referred To As
1006	U.S. Patent 5,797,119	Ozawa
1004	Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP)	G.729
1007	U.S. Patent 5,235,669	Ordentlich

A. Ozawa

83. Ozawa discloses solving the problem of the quality of female speech degrading significantly due to the limited codebook size associated with low coding rates by disclosing “a low bit rate speech coding technique that allows reduction of computations associated with a comb filtering process and provides immunity to bit errors.” Ex. 1006, 1:63-65.

84. In light of the use of various terms having the same meaning throughout the '521 Patent and the prior art, I provide the following chart illustrating synonymous terms used throughout some of the references cited herein.

'521 Patent Terms (Ex. 1001)	Ozawa Terms (Ex. 1006)
Pitch codebook	Adaptive codebook
Pitch codevector	Adaptive codevector/pitch period
Pitch prediction error calculating device	Selector or Gain calculation
Target vector	Corrected speech signal $X'w(n)$
Filter index	Mode index
Pitch codebook index	Pitch index or excitation index
Pitch gain	Gain index
Innovative codebook	Excitation codebook
Innovative search target vector	Excitation codebook vector
Pitch prediction error	Distortion
Energy of pitch prediction error	Sum squared distortion

85. Some of the challenged claims (e.g., claims 6, 8, 15, 17, 33, 42, 44) contain terms regarding devices or means for calculating quantities related to the constructions proposed above. In relation to these terms, Ozawa discloses functional elements that implement signal processing algorithms expressed by mathematical equations (e.g., equations 1-11). In addition, Ozawa explains that the disclosed apparatus performs filtering and searching methods for high quality speech coding that require “large amount of computations.” Ex. 1006, 1:53-56. For example, Ozawa implements linear predictive coding (LPC) where “[c]omputations are performed on the signals extracted through the window to produce a spectral parameter of the speech samples.” Ex. 1006, 2:65-67. In

addition, the search for minimum distortion requires “computations involved in the gain calculation, excitation pulse syntheses and impulse response calculation on excitation pulses.” See Ex. 1006, 6:16-20. A person of ordinary skill in the art would understand that the speech coder of Ozawa necessarily incorporates a processor configured to perform the various computations associated with respective functional elements in accordance with the respective mathematical equations disclosed by Ozawa.

1. Claims 1, 2, 5, 7, 8, 55, 56, 59, and 62 Are Anticipated by Ozawa

a. Claim 1

86. Claim 1 of the '521 Patent recites:

1. A pitch analysis device for producing a set of pitch codebook parameters, comprising:
a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;
a) at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, wherein:
 - 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; and
 - ii) 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; and

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.

87. Ozawa discloses a pitch codebook search device to generate a pitch code vector based on digitized input audio data, which is incorporated into pitch synthesizer 17 as described in Ex. 1006, 4:17-26. The adaptive code book in the pitch synthesizer 17 provides adaptive code vectors (pitch code vectors), which are the pitch contribution to the excitation signal according to '521 Patent at 1:54-59. Ex. 1006, 5:22-43. The pitch index of Ozawa is used to select a pitch code vector from the adaptive code book in the pitch synthesizer 17 of Ozawa, and thus provides the pitch code vector corresponding to the '521 Patent.

88. Accordingly, Ozawa discloses “[a] pitch analysis device for producing a set of pitch codebook parameters comprising: a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data,” as required by claim 1.

89. Ozawa discloses digitized input audio data. The input signal in Ozawa Fig. 1 is identified as “digital speech input” which is by definition digitized audio data. Specifically, Ozawa discloses that “an input speech signal is segmented into speech samples” Ex. 1006, 1:66-2:7.

90. Accordingly, Ozawa discloses “said digitized input audio data represents an input audio signal that has been sampled and digitized,” as required by claim 1.

91. Ozawa discloses at least two signal paths associated with respective sets of pitch codebook parameters representative of the digitized input audio data. One signal path employs filtered code vectors (modes 1-3) and the second signal path employs excitation vectors without filtering (mode 0). Ex. 1006, 6:21-40. Fig. 2 shows these two sets of signal paths coming from the pitch synthesizer 17 and the excitation vector candidate selection 18. The first set of signal paths carries the output of the excitation vector candidate selection 18 through filter 21, while the other signal path routes the output of the excitation vector candidate selection 18 directly to the selector 22, bypassing filter 21.

92. Accordingly, Ozawa discloses “at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data,” as required by claim 1.

93. Ozawa also discloses a pitch prediction error calculating device, for calculating the pitch prediction error of the pitch codevector from the pitch codebook search device. In Ozawa, this calculating device is vector selector 22, which calculates distortion (prediction error) D of the pitch codevector in equation (5). Ex. 1006, 5:8-21, using the equation

$$D_k = \sum_{n=0}^{N-1} \left[X'w(n) - \beta'_k q(n) - g'_{1k} c_{1jz}(n) * h_w(n) - g'_{2k} c_{2kz}(n) * h_w(n) \right]^2 \quad (\text{Ex. 1006, eq. (6)}),$$

where $X'w(n)$ is the corrected speech sample target, c_{1jz} and c_{2kz} are code vectors, g'_{1k} and g'_{2k} are code vector gains, $h_w(n)$ is an impulse response, and β'_k and $q(n)$ are the k^{th} adaptive code vector pitch gain and pitch index. This formula would be immediately recognizable to one of skill in the art as a calculation of sum squared prediction error which includes the long-term prediction component given by the pitch gain and pitch index. This indicates that Ozawa's distortion calculation is necessarily a pitch prediction error calculation.

94. Further, Ozawa discloses that “[i]n each operating mode, the gain calculator generates a gain index representing the quantized optimum gain code vector” Ex. 1006, 5:40-41. This represents a separate error calculation and optimization for each operating mode, where the operating modes 1, 2, and 3 constitute signal paths that include a filter, while in mode 0 there is no filter.

95. Accordingly, Ozawa discloses “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,” as required by claim 1.

96. Ozawa discloses that at least one of the signal paths includes a filter. In Ozawa the signal path corresponding to modes 1, 2, and 3 comprises filter 21 for filtering the pitch code vector before supplying it to the pitch prediction error calculating device of the one signal path. In Ozawa, the filter is specified as a comb filter 21 that receives the excitation vector candidates, the pitch index, and the

mode index. Specifically, “[d]uring mode 1, 2, or 3, each of the excitation code vector candidates is passed through the comb filter”. (Ex. 1006, 4:57-61). Thus, the comb filter 21 filters pitch codevectors specified by the pitch index. Since Ozawa specifies that “[a]lternatively, a different value of weighting function p may be used for each mode of operation” (EX. 1006, 4:67-5:1), the comb filter may be considered as one signal path and one comb filter, or alternatively as three signal paths with three different comb filters, depending on the specific implementation.

97. Accordingly, Ozawa discloses “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 required by claim 1.

98. Ozawa discloses a selector for comparing the pitch prediction errors calculated in the signal paths and for choosing the signal path and corresponding pitch codebook parameters having the lowest calculated pitch prediction error. Specifically, “[d]uring mode 1, 2 or 3, vector selector 22 searches for a gain code vector that minimizes Equation (6) with a respect to each of the filtered excitation code vectors, and during mode 0 it searches for one that minimizes Equation (7) with respect to each excitation code vector. Vector selector 22 selects one of the candidate code vectors and one of the gain code vectors that minimize the distortion given Equation (6) during mode 1, 2 or 3, or minimize the distortion

given by Equation (7) during mode 0, and delivers the selected candidate excitation code vector and the selected gain code vector as excitation and gain indices to multiplexer 25 as well as to excitation pulse synthesizer 28.” Ex. 1006, 6:21-40.

99. 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. The modes, controlled by specified ranges of the overall error, are each designed with pitch prediction filter characteristics (or lack of a filter, in the case of mode 0) most well-suited to minimize error within that specific prediction error range, customizing the signal paths for that purpose. 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.

100. Regardless of operating mode, all required codebook parameters are optimized to minimize prediction error and delivered to the multiplexor 25. “Vector selector 22 selects one of the candidate code vectors and one of the gain code vectors that minimize the distortion given Equation (6) during mode 1, 2 or 3, or minimize the distortion given by Equation (7) during mode 0, and delivers the selected candidate excitation code vector and the selected gain code vector as excitation and gain indices to multiplexer 25 as well as to excitation pulse synthesizer 28.” (Ex. 1006, 6:21-40)

101. Accordingly, Ozawa discloses “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,” as required by claim 1.

102. Accordingly, Ozawa discloses all of the limitations of claim 1.

b. Claim 2

103. Claim 2 of the '521 Patent recites:

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.

104. Ozawa discloses “[a] pitch analysis device as defined in claim 1,” as shown above in ¶¶87-102.

105. Ozawa also discloses that one signal path comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device. In Ozawa the signal path comprising no filter is shown clearly in Ex. 1006, Fig. 2, as the signal lines bypassing the filter and going directly into selector 22. “During mode 1,2 or 3. each of the excitation code vector candidates is passed through the comb filter” (Ex. 1006, 4:59-61), whereas the set of signal paths comprising no filter is disclosed as being associated with the designated mode 0, which is associated with unvoiced signals, those “in which the

pitch synthesizer 17 is producing no outputs” (Ex. 1006, 4:43-44). See Ex. 1006 6:21-33 .

106. Accordingly, Ozawa discloses “[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,” as required by claim 2.

c. Claim 5

107. Claim 5 of the ’521 Patent recites:

5. A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:

- a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

108. Ozawa discloses “[a] pitch analysis device as defined in claim 1,” as shown above in ¶¶87-102.

109. Ozawa discloses a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector.

110. In Ozawa, convolution between the pitch codevector and the weighted synthesis filter impulse response is disclosed in eq. (10).

$$d(n) = v(n) - \sum_{l=1}^{10} \alpha_l \cdot q(n-l) + \sum_{l=1}^{10} \alpha_l \cdot g^l \cdot p(n-l) + \sum_{l=1}^{10} \alpha'_l \cdot g^l \cdot d(n-l) \quad (10)$$

where $q(n)$ is the pitch codevector, $p(n)$ is the weighting filter, and the α_l and α'_l terms are the unquantized and quantized spectral parameters (Ex. 1006, 5:56-67) that expressly characterize the filter impulse response. (Ex. 1006, 1:57-59). As well known to one of ordinary skill in the art, convolution is a standard mathematical formula that describes the output signal of a filter, e.g. $y = x * h$, where x is the input signal, h is the filter impulse response, $*$ denotes convolution, and y is the resulting output signal from the filter (Ex. 1035, Oppenheim and Schaefer 1989). The difference equation (10) directly corresponds to the z-transform transfer function and corresponding impulse response for the implemented perceptually weighted synthesis filter, which is convolved with $v(n)$, the sum of the pitch codevector and the excitation codevectors. This convolution

equation is disclosed as being implemented within the excitation pulse synthesizer unit 28.

111. Accordingly, Ozawa discloses “a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector,” as required by claim 5.

112. Ozawa discloses a gain calculator 23 that searches a codebook 24 for a gain code vector that includes the pitch gain parameter β'_k , which is the gain of k-th adaptive code vector. Specifically, Ozawa discloses “The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation:

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

where, β'_k is **the gain of k-th adaptive code vector.**”_Ex. 1006, 5:22-32 (emphasis added).

113. Accordingly, Ozawa discloses “a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector,” as required by claim 5.

114. Ozawa discloses an amplifier for multiplying the convolved pitch codevector by the pitch gain. In Ozawa the amplifier is implemented via a multiplier in Equation (6). Ozawa discloses in equation 6 that the k-th adaptive code vector (pitch code vector) is multiplied by the gain parameter β'_k . Thus, Ozawa discloses a multiplier as an amplifier for multiplying the convolved pitch code vector $q(n)$ to obtain an amplified convolved pitch code vector.

115. Accordingly, Ozawa discloses “an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector,” as required by claim 5.

116. Ozawa discloses a pitch search target vector, denoted the corrected speech signal $X'w(n)$. “In order to determine the pitch period at subframe intervals. the pitch synthesizer 17 is provided with a known adaptive codebook. During mode 1, 2, or 3. a pitch period is derived from an output sample $X'w(n)$ of subtractor 16 using the impulse response $hw(n)$ from impulse response calculator 26.” (Ex. 1006, 4:17-22). The corrected speech signal $X'w(n)$ is the difference between the perceptual weighted speech $Xw(n)$ and a correcting signal $Xz(n)$ in Fig. 1. The pitch codevector is identified in the pitch synthesizer 17 and produces as the convolved pitch codevector $q(n)$, as shown in Fig. 1 (Ex. 1006, Fig. 1), which is subtracted from the target vector $X'w(n)$.

117. Ozawa also discloses amplification of the convolved pitch codevector $q(n)$, through the gain parameter β in equations (2), (5), and (6). “... β is the gain of the pitch synthesizer 17.” Ex. 1006, 4:38-39.

118. Ozawa discloses a combiner circuit that combines the amplified convolved pitch code vector, represented by the pitch index $q(n)$ amplified by the parameter β , with the excitation vector selected by excitation vector candidate selector 18 to calculate the distortion (i.e. pitch prediction error) in equations (2), (5), and (6). Each of these equations explicitly discloses the combination of the convolved pitch code vector and the selected excitation vectors through addition. Specifically, “[t]he output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation: ... ” (Ex. 1006, 5:22-30). Each of the excitation vector candidate selection 18, the selector 22, and the gain calculation 23 incorporates a combiner circuit that calculates corresponding distortion expressions, *e.g.*, equations 2, 5, and 6. See Ex. 1006, 5:44-6:20; 3:23-31. 4:17-26.

119. Accordingly, Ozawa discloses “a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error,” as required by claim 5.

120. Accordingly, Ozawa discloses all of the limitations of claim 5.

d. Claim 7

121. Claim 7 of the '521 Patent recites:

- 7. A pitch analysis device as defined in claim 5, wherein:
 - a) each of said filters of the plurality of signal paths is identified by a filter index;
 - b) said pitch codevector is identified by a pitch codebook index;
 - c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.

122. Ozawa discloses “[a] pitch analysis device as defined in claim 5,” as shown above in ¶¶108-120.

123. As discussed in ¶96 for claim 1, Ozawa discloses one or more signal paths that include a comb filter, as well as a signal path that does not include a comb filter. Ozawa discloses a mode index which acts as a filter index to determine which signal path is selected, being either mode 1, 2, or 3, in which excitation vector candidates are put through a comb filter, or mode 0, in which excitation vector candidates are not put through the filter. In Ozawa, the mode index represents a particular selection of the weighting function ρ (rho) applied by the comb filter 21, which thereby creates a potentially unique filter characteristic for each mode index, depending on the values of ρ . (Ex. 1006, 4:67-5:2).

124. Modes 1, 2, and 3 act as a filter index to select the filter path, which may be three different paths in the alternative case where the filter weighting function is

different for each mode, while mode 0 acts as a filter index to select the non-filter path. Thus, each of the one or more filters of the plurality of signal paths is identified by a filter index, as is the non-filter path. Thus, the mode index operates as a filter index that selects a different value of the weighting function ρ .

125. Accordingly, Ozawa discloses “each of said filters of the plurality of signal paths is identified by a filter index,” as required by claim 7.

126. Ozawa discloses a pitch codebook index, which it refers to as the pitch index (Ex. 1006, 4:22-26). Ozawa explains that the “[p]itch synthesizer 17 supplies a **pitch index indicating the pitch period to an excitation vector candidate selector 18**, a comb filter 21, a vector selector 22, an excitation pulse synthesizer 28 and to the multiplexer 25. When the encoder is in mode 0, the pitch synthesizer produces no pitch index.” Ex. 1006, 4:22-26 (emphasis added).

127. Accordingly, Ozawa discloses “said pitch codevector is identified by a pitch codebook index,” as required by claim 7.

128. Ozawa discloses that the “[e]xcitation pulse synthesizer 28 receives the gain index, excitation indices, mode index and pitch index and reads corresponding vectors from codebooks.” Ex. 1006, 5:44-55.

129. Accordingly, Ozawa discloses “said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain,” as required by claim 7.

130. Accordingly, Ozawa discloses all of the limitations of claim 7.

e. Claim 8

131. Claim 8 of the '521 Patent recites:

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.

132. Ozawa discloses “[a] pitch analysis device as defined in claim 1,” as shown above in ¶¶87-102.

133. As discussed above in ¶¶93-94 for the pitch prediction error calculating device of claim 1, Ozawa discloses a vector selector 22 that calculates a distortion D which is calculated as a sum of squares of the components of the prediction error vector using equation 5. (Ex. 1006, 5:8-21). Specifically, Ozawa discloses that “[t]he filtered vector candidates $C_{jz}(n)$, the pitch index and the output of subtractor 16 are applied to the vector selector 22. For the first and second excitation vector stages (corresponding respectively to codebooks 19 and 20), the vector selector 22 selects those of the filtered candidates which **minimizes the distortion given by the following Equation:**

$$D_c = \sum_{n=c}^{N-1} |Xw(n) - \beta \cdot q(n) - g_1c_{1z}(n) * h_w(n) - g_2c_{2z}(n) * h_w(n)|^2 \quad (5)$$

25.” Ex. 1006, 5:8-21 (emphasis added).

134. The distortion or sum-squared value of the pitch prediction error given by this equation is synonymous with the energy of the pitch prediction error, as understood by a person of ordinary skill in the art.

135. Ozawa also discloses that “[t]he output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector **that minimizes distortion represented by the following Equation:**

$$D_k = \sum_{n=c}^{N-1} [X^*w(n) - \beta_k \cdot q(n) - g'_{11}c_{11}(n) \cdot h_w(n) - g'_{22}c_{22}(n) \cdot h_w(n)]^2 \quad (6)$$

Id. 5:22-39 (emphasis added).

136. The gain calculator 23 calculates the distortion D, the sum of the squares of the components of the prediction error vector per equation 6. Ex. 1006, 5:22-39. The distortion expressed in each of equations 5 and 6 takes into account the contribution of the pitch vector (e.g. “k-th adaptive code vector,” Ex. 1006, 5:32) and is thus a pitch prediction error. The distortion or sum-squared value of the pitch prediction error given by equations 5-6 is the energy of the pitch prediction error, as understood by a person of ordinary skill in the art. Since equations 5-6 involve multiplication and subtraction operations, gain calculator 23 and vector

selector 22 are necessarily configured to perform multiplications and subtraction operations to calculate the energy of the pitch prediction error. Thus, Ozawa discloses gain calculator 23 and vector selector 22 as a circuit configured to use a multiplier and a subtractor to calculate the energy of the pitch prediction error.

137. Accordingly, Ozawa discloses “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” as required by claim 8.

138. As discussed above, Ozawa discloses minimizing distortion. See Ex. 1006 5:8-21 and 5:22-39, above (“minimizes distortion”).

139. Accordingly, Ozawa discloses “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,” as required by claim 8.

140. Accordingly, Ozawa discloses all of the limitations of claim 8.

f. Claim 55

141. Claim 55 of the '521 Patent recites:

55. A pitch analysis method for producing a set of pitch codebook parameters, comprising:
generating a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;

- a) in at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, calculating, for each signal path, a pitch prediction error of said pitch codevector from said pitch codebook search device;
- b) in at least one of said at least two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path;
- c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.

142. Claim 55 recites the method form of claim 1. The pitch analysis device for producing a set of pitch codebook parameters of claim 1 discussed above performs the method steps recited in claim 55.

143. Ozawa discloses “[a] pitch analysis method for producing a set of pitch codebook parameters,” as shown above in ¶¶87-88.

144. Ozawa discloses “generating a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized,” as shown above in ¶¶87-90.

145. Ozawa discloses “in at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, calculating, for each signal path, a pitch prediction error of said pitch codevector from said pitch codebook search device,” as shown above in ¶¶91-95.

146. Ozawa discloses “in at least one of said at least two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path,” as shown above in ¶¶96-97.

147. Ozawa discloses “comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path,” as shown above in ¶¶98-101.

148. Accordingly, Ozawa discloses all of the limitations of claim 55.

g. Claim 56

149. Claim 56 of the '521 Patent recites:

56. A pitch analysis method as defined in claim 55, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device.

150. Claim 56 is the same as claim 2 except that it depends from claim 55.

151. Ozawa discloses “[a] pitch analysis device as defined in claim55,” as shown above in ¶¶143-148.

152. Ozawa discloses “wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device,” as shown above at ¶¶105-106.

153. Accordingly, Ozawa discloses all of the limitations of claim 56.

h. Claim 59

154. Claim 59 of the '521 Patent recites:

59. A pitch analysis method as defined in claim 55, wherein calculating a pitch prediction error in each signal path comprises:

- a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

155. Claim 59 is similar to claim 5 except that it depends from method claim 55.

156. Ozawa discloses “[a] pitch analysis device as defined in claim 55,” as shown above in ¶¶143-148.

157. Ozawa discloses “wherein calculating a pitch prediction error in each signal path comprises: a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector,” as shown above in ¶¶109-111.

158. Ozawa discloses “calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector,” as shown above in ¶¶112-113.

159. Ozawa discloses “multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector,” as shown above in ¶¶114-115.

160. Ozawa discloses “combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error,” as shown above in ¶¶116-119.

161. Accordingly, Ozawa discloses all of the limitations of claim 59.

i. Claim 62

162. Claim 62 of the '521 Patent recites:

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.

163. Claim 62 is similar to claim 8 except that it depends from method claim 55.

164. Ozawa discloses “[a] pitch analysis device as defined in claim 55,” as shown above in ¶¶143-148.

165. Ozawa discloses “wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error,” as shown above in ¶¶133-137.

166. Ozawa discloses “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,” as shown above in ¶¶138-139.

167. Accordingly, Ozawa discloses all of the limitations of claim 62.

B. G.729 and Ozawa

168. Ozawa discloses that “the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by” Equation 6. Ex. 1006, 5:24-30. As discussed in ¶¶171-175 below, G.729 discloses this exact equation for calculating pitch gain. It would have been obvious to a person of ordinary skill in the art to configure the gain calculator 23 of Ozawa to calculate the pitch gain using the equation for adaptive-gain g_p disclosed in G.729, which involves calculating the pitch gain using multiplication, subtraction, and division operations. Thus, Ozawa in view of G.729 discloses a gain calculator 23 incorporating “a multiplier, a subtractor, and a division unit to calculate the pitch gain using the equation set forth at 12:40-43.”

2. Claims 6 And 60 Are Obvious Over Ozawa In View Of G.729

a. Claim 6

169. Claim 6 of the '521 Patent recites:

6. A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

170. Ozawa discloses “[a] pitch analysis device as defined in claim 5,” as shown above in ¶¶108-120.

171. G.729 discloses this formula calculating the filter gain. Ex. 1004, 19.

172. It was known in the art that pitch gain can be calculated using the relation $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is the pitch search target vector and $y^{(j)}$ is the convolved pitch codevector.

173. G.729 discloses that the adaptive-gain g_p is calculated using the following formula:

$$g_p = \frac{\sum_{n=0}^{39} x(n) y(n)}{\sum_{n=0}^{39} y(n) y(n)} \quad \text{bounded by } 0 \leq g_p \leq 1.2$$

174. G.729 explains that $x(n)$ is the target signal and $y(n)$ is the filtered adaptive-codebook vector. Ex. 1004, at 19. It is further explained that vector $y(n)$ is obtained by convolving $v(n)$ and $h(n)$ using the following equation:

$$y(n) = \sum_{i=0}^n v(i)h(n-i) \quad n = 0, \dots, 39$$

Ex. 1004. As can be seen from the equation for g_p , this is the same relation required in claim 6 where g_p of G.729 is the same as $b^{(j)}$. The equation also uses a multiplier and a subtractor to calculate the mean squared prediction error.

175. Accordingly, G.729 discloses “wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector,” as required by claim 6.

176. It would have been obvious to one of skill in the art to combine G.729 with Ozawa. Ozawa discloses gain calculator 23. G.729 discloses the formula for calculating the gain. It would have been obvious for a person of ordinary skill in the art to calculate the adaptive-codebook gain (pitch gain) using equation 43 as provided in G.729.

177. G.729 codifies a CELP encoder and decoder structure for coding of speech signals at 8 kbits/s. G.729 uses algebraic CELP (ACELP), which is a specific

implementation of CELP that maintains all general structure of the CELP in the same form that has been used in the art since the early days of CELP. *See* Ex. 1004, 3-4, 8 (Figs. 2, 3, and 4); Ex. 1008, Figs. 1-2 (U.S. Pat. No. 5,444,816 to Adoul et al. is cited as admitted prior art in the '521 patent, 13:11-14). Ozawa also discloses a CELP codec that adopts the same generic form for CELP except for additional functional blocks. *See* Ex. 1006, Figs. 1-2. Although Ozawa introduces a comb filter 21 to optionally filter the excitation vector candidates, the principle of operation of Ozawa's CELP codec remains substantially the same as the generic codec also implemented in the G.729 standard. In fact, G.729 also uses a "filtered adaptive-codebook vector" in its computations. Ex. 1004, 19.

178. Ozawa discloses that the pitch gain β_k associated with the convolved pitch codeword $q(n)$ is identified using a codebook search in a three dimensional gain codebook 23 that includes the adaptive codevector gain as well as the two fixed codebook gains. This is a more complex approach to gain calculation than the direct computation of the '521 patent or equivalently of G.729, for the purpose of reducing the number of transmission bits. It would have been obvious to one skilled in the art to apply the express teaching of G.729 to simplify the Ozawa system through separation of the adaptive and fixed codebook gains and direct implement of the adaptive gain calculation using the teaching of G.729. A person of ordinary skill in the art would also have known that the established principle for

calculating the gain of adaptive code vector in the context of CELP coding is through Equation 43 as codified in G.729. *Compare* Ex. 1004 (Equation 43) with Ex. 1008, 8:20-24 (U.S. 5,444,816 to Adoul et al., Equation 7); Ex. 1028, 9:25-43 (U.S. Patent No. 5,293,449 to Tzeng, Equation 9); Ex. 1029, 16:56-68 (U.S. Pat. No. 4,868,867 to Davidson, Equation 10). Thus, claim 6 of the '521 Patent simply claims a known mathematical formula for calculating the gain of a codevector.

179. Accordingly, Ozawa in view of G.729 discloses all of the limitations of claim 6.

b. Claim 60

180. Claim 60 of the '521 Patent recites:

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:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

181. Claim 60 is the same as claim 6 except that it depends from claim 59.

182. Ozawa discloses “[a] pitch analysis method as defined in claim 59,” as shown above in ¶¶156-161.

183. G.729 discloses “wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation, $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search

target vector and $y^{(i)}$ is said convolved pitch codevector,” as shown above in ¶¶171-175.

184. It would have been obvious to one of skill in the art to combine G.729 with Ozawa as shown above in ¶¶176-178.

185. Accordingly, Ozawa in view of G.729 discloses all of the limitations of claim 60.

C. Ordentlich

1. Claims 10, 11, 14, 17, 28, 29, 32, 35, 37, 38, 41, And 44 Are Obvious Over Ozawa In View Of Ordentlich

a. Claim 10

186. Claim 10 of the '521 Patent recites:

10. An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:

a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;

b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;

c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;

d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:

i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector

and an innovative search target vector; and

- 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;
- e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters;
- f) a signal forming device for producing an encoded wideband 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.

187. Ozawa discloses “[a]n encoder having a pitch analysis device as in claim 1,” as shown above in ¶¶87-102.

188. Ordentlich discloses coding of wideband speech using a CELP encoder. Ex. 1007, 1:64-2:2. Ordentlich states that it “seeks to adapt existing CELP techniques to extend to communication of such **wide-band speech** and other such signals.” Ex. 1007, 1:64-2:2 (emphasis added). Ordentlich also discloses that “wideband speech is assigned the band 50 to 7000 Hz and is sampled at a rate of 16000 Hz for subsequent digital processing.” Ex. 1007, 1:41-54. One of skill in the art would understand that sampling and digital processing of wideband speech by definition indicates a wideband input signal.

189. Ozawa discloses a linear prediction synthesis filter calculator for producing linear prediction filter coefficients. Specifically Ozawa discloses “In the LSP

calculation circuit 12, linear predictive coefficients α_i , (where i corresponds to the order p and equals to 1, 2 ... 10) which are obtained by the Burg's method are converted to linear spectrum pairs. Or LSP parameters suitable for quantization and interpolation processes.” Ex. 1006, 3:13-18. Calculation of the linear prediction coefficients takes place in the LSP calculation circuit 12 using the well-known Burg’s method (Burg 1975).

190. Accordingly, Ozawa discloses “a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients,” as required by claim 10.

191. Ozawa also discloses a perceptual weighting filter, which receives the wideband signal and linear prediction synthesis filter coefficients. This can be seen in Fig. 1 where the wideband signal and filter coefficients from spectral parameter calculation unit 12 are provided to perceptual weighting circuit 15. Perceptual weighting circuit 12 produces weighted digital speech samples $X_w(n)$. Ex. 1006, at 3:19-25, *see also* Fig. 1.

192. Ozawa discloses that “[t]he linear predictive coefficients α_{ij} (where j indicates subframes 1 to 5) are supplied at subframe intervals from the circuit 12 **to the perceptual weighting circuit_15** so that the speech samples from the subframing circuit 11 are weighted by the linear predictive coefficients. A series of

perceptually weighted digital speech samples $X_w(n)$ are generated and supplied to a subtractor 16 . . .” Ex. 1006, 3:19-25 (emphasis added).

193. Accordingly, Ozawa discloses “a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal,” as required by claim 10.

194. Ozawa further discloses an impulse response generator responsive to the linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal. The impulse response generator of Ozawa is the impulse response calculator 26 shown in Fig. 1. According to Ozawa, the impulse response calculator uses the linear predictive coefficients and calculates impulse responses $h_w(n)$. Ex. 1006, 3:56-67. This is supplied to the pitch synthesizer 17 to allow it to determine the pitch period of the speech signal.

195. Ozawa specifically discloses that “[t]he **linear predictive coefficients** (where j indicates subframes 1 to 5) are supplied at subframe intervals from the circuit 12 **to the perceptual weighting circuit** 15 so that the speech samples from the subframing circuit 11 **are weighted by the linear predictive coefficients**. Ex. 1006, 3:19-23(emphases added).

196. Accordingly, Ozawa discloses “an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal,” as required by claim 10.

197. Ozawa also discloses a pitch search unit for producing pitch codebook parameters. Specifically, Ozawa states that “[i]n order to determine the pitch period at subframe intervals, the pitch synthesizer 17 is provided with **a known adaptive codebook**. During mode 1, 2 or 3, a pitch period is derived from an output sample $X'w(n)$ of subtractor 16 using the impulse response $h_w(n)$ from impulse response calculator 26. Pitch synthesizer 17 **supplies a pitch index indicating the pitch period**” Ex. 1006, 4:17-23 (emphases added).

198. The adaptive codebook in the pitch synthesizer 17 provides adaptive code vectors or pitch periods, which are the pitch code vectors of the '521 Patent. Ex. 1001, 4:17-26. The excitation codebooks 19 and 20 are innovative codebooks that provide excitation codebook vectors, which are the innovative search target vectors of the '521 Patent. Ex. 1001, 4:27-53.

199. Accordingly, Ozawa discloses “a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising: i) said pitch codebook search device responsive to the perceptually weighted signal and the linear prediction synthesis filter coefficients for producing the pitch codevector,” as required by claim 10.

200. Ozawa also discloses a device for selecting the innovation (excitation) code vectors and the pitch codebook parameters having lowest pitch prediction error. Ozawa states “[t]he output of the vector selector 22, the pitch index from pitch

synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation” Ex. 1006, 5:22-32.

201. Accordingly, Ozawa discloses “an innovative search target vector; and 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,” as required by claim 10.

202. Ozawa discloses an innovative codebook search device. Ozawa states that “[e]xcitation vector candidate selector 18 is connected to excitation vector codebooks 19 and 20 to search for excitation vector candidates and **to select excitation code vectors such that those having smaller amounts of distortion**

$$D = \sum_{n=0}^{N-1} |Xw(n) - \beta \cdot q(n) - g_1 \cdot c_{1f}(n) * h_w(n) - g_2 \cdot c_{2f}(n) * h_w(n)|^2 \quad (2)$$

are selected with higher priorities. When the encoder is in mode 1, 2 or 3, it makes **a search through the codebooks 19 and 20 for excitation code vectors that minimize the amount of distortion represented by the following Equation:** ”

Ex. 1006 4:27-37 (emphases added).

203. Ozawa discloses that the excitation vector candidate selection 18 is responsive to the impulse response signal because it uses the parameters $h(n)$ to calculate the prediction error. Ex. 1006 4:27-53, Equations 2-3 and Fig. 1.

204. Accordingly, Ozawa discloses “an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters,” as required by claim 10.

205. Ozawa does not explicitly disclose that the encoded signal is wideband but as discussed above, this is disclosed in Ordentlich. Ozawa does, however, disclose a signal forming device. In Ozawa, multiplexer 25 is the signal forming device that generates coded information for transmission, comprising the pitch codebook parameters associated to the signal path having the lowest pitch prediction error, the innovative codebook parameters, and the linear prediction synthesis filter coefficients. Ex. 1006, 3:48-55 (“coefficients are supplied to the multiplexer 25”); , 4:15-17 (“mode index is supplied multiplexer 25”); 4:22-25 (“pitch index indicating the pitch period is supplied to multiplexer 25”); 5:18-21 (“excitation indices supplied to multiplexer 25”); 5:40-43 (“gain index supplied to multiplexer 25”).

206. Accordingly, Ozawa discloses “a signal forming device for producing an encoded wideband 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,” as required by claim 10.

207. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich. The CELP codec of Ozawa is a typical CELP implementation that would easily generalize to signals of arbitrary bandwidth. *See* Ex. 1006, Figs. 1-2. Although Ozawa introduces a comb filter 21 to optionally filter the excitation vector candidates (equivalent to the multiple signal paths of the ‘521 Patent), the principle of operation of Ozawa’s CELP codec remains substantially the same as the generic CELP codec also implemented in Ordentlich.

208. Ordentlich discloses a low-delay code-excited linear-predictive coding of wideband speech. Similar to Ozawa, Ordentlich discloses an interconnection of conventional CELP components (*see* Ex. 1007, 2:37-40, and Fig. 1) including an excitation codebook 10 that provides a codevector (innovation codevector) that is combined with “segments of past excitation” (pitch codevector) to perform pitch search. *See* Ex. 1007, 2:43-49; 3:26-30. Ordentlich discusses having a different number of taps/bits for the pitch filter. Ex. 1007, 7:54-53-55. Also, just like Ozawa, Ordentlich discloses that “it sometimes proves advantageous to use a system without a pitch loop, i.e., $B(z)=1$.” Ex. 1007, 7:56-57.

209. Ordentlich discloses inputting a wideband speech signal having a frequency bandwidth of 50 to 7000 Hz to the CELP system. Ex. 1007, 5:40-42. Thus, Ordentlich expressly includes a suggestion to modify a CELP encoder to encode wideband speech. For example, Ordentlich discloses that the wideband speech can be directly applied to the CELP codec by providing a filter $W(z)$ that provides perceptual quality. Specifically, Ordentlich states that “The filter $W(z)$ is important for achieving a high perceptual quality in CELP systems and it plays a central role in the CELP-based wideband coder presented here.” Ex. 1007, 3:22-25, and discusses controlling quantization noise in different frequencies, stating “On the other hand, the auditory system is quite sensitive in this region and the quantization distortions are clearly audible in a form of crackling and hiss. Noise weighting is, therefore, more crucial, in wideband CELP. The balance of low to high frequency coding is more delicate. The major effort in this study was towards finding a good weighting filter that would allow a better control of this balance.” Ex. 1007, 5:59-66. Just like Ordentlich, Ozawa discloses using a perceptual weighting filter 15.

210. A person of ordinary skill in the art would consider that the arrangement of Ordentlich is compatible with the arrangement of Ozawa and would be motivated to apply a wideband speech at the input of Ozawa as expressly suggested by Ordentlich.

211. A person of ordinary skill in the art would have been motivated to follow the teachings of Ordentlich to adapt the existing CELP techniques disclosed in Ozawa to the coding of wideband speech signals. Ordentlich discloses positive benefits that a person of ordinary skill in the art would understand of signs of success for using the teachings of Ordentlich. For example, Ordentlich discloses that “[t]he added low frequencies enhance the naturalness and authenticity of the speech sounds. The added high frequencies make the sound crisper and more intelligible.” Ex. 1007, 5:42-47.

212. A person of ordinary skill in the art would thus have been motivated by the teachings of Ordentlich to apply the coding method of Ozawa to wideband speech.

213. Thus, adding a wideband speech signal at the input of a CELP encoder to produce an encoded wideband signal was already known in the art as exemplified by Ordentlich. The expected result of such a combination as disclosed by Ozawa is a wideband encoded speech signal with added high frequencies that make the sound crisper and more intelligible, and a better control of the balance of low to high frequency coding (Ex. 1006).

214. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 10.

b. Claim 11

215. Claim 11 of the '521 Patent recites:

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.

216. Claim 11 is similar to claim 2 except that it depends from claim 10.

217. Ozawa in view of Ordentlich discloses “[a]n encoder as defined in claim 10,” as shown above in ¶¶187-214.

218. Ozawa discloses “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 shown above in ¶¶105-106.

219. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

220. Accordingly, Ozawa in view of Ordentlich discloses all the limitations of claim 11.

c. Claim 14

221. Claim 14 of the '521 Patent recites:

14. An encoder as defined in claim 10, wherein each pitch prediction error calculating device comprises:
a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

222. Claim 14 is similar to claim 5 except that it depends from claim 10.

223. Ozawa in view of Ordentlich discloses “[a]n encoder as defined in claim 10,” as shown above in ¶¶187-214.

224. Ozawa discloses “wherein each pitch prediction error calculating device comprises: a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector,” as shown above in ¶¶109-111.

225. Ozawa discloses “a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector,” as shown above in ¶¶112-113.

226. Ozawa discloses “an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector,” as shown above in ¶¶114-115.

227. Ozawa discloses “a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error,” as shown above in ¶¶116-119.

228. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

229. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 14.

d. Claim 17

230. Claim 17 of the '521 Patent recites:

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.

231. Claim 17 is similar to claim 8 except that it depends from claim 10.

232. Ozawa in view of Ordentlich discloses “[a]n encoder as defined in claim 10,” as shown above in ¶¶187-214.

233. Ozawa discloses “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” as shown above in ¶¶133-137.

234. Ozawa discloses “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,” as shown above in ¶¶138-139.

235. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

236. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 17.

e. Claim 28

237. Claim 28 of the '521 Patent recites:

28. A cellular mobile transmitter/receiver unit, comprising:
a) 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;
and b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.

238. Ordentlich discloses a cellular mobile transmitter/receiver unit. Ordentlich discloses that “[o]ne area of such improvements have come to be called code excited linear predictive (CELP) coders . . . Such techniques have found

application, e.g., in voice grade telephone channels, including **mobile telephone channels.**” Ex. 1007, 1:25-40 (emphasis added).

239. Ozawa in view of Ordentlich discloses “a transmitter including an encoder for encoding a wideband signal as recited in claim 10,” as shown above in ¶¶187-214.

240. Ozawa discloses a transmissions circuit. According to Ozawa, “[i]n each operating mode. the gain calculator generates a gain index representing the quantized optimum gain code vector for application to the excitation pulse synthesizer 28 as well as to the multiplexer 25 **for transmission** or storage.” Ex. 1006, 5:40-44

241. Ordentlich discloses a transmission circuit for transmitting the encoded wideband signal. Ordentlich discloses that “[s]hown are the transmitter portion at the top of the figure, the receiver portion at the bottom and the various parameters (j , g , M , β and A) that are transmitted via a communication channel 50. “ Ex. 1007, 2:40-43. And, that “[i]n general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. The transmitted information is used at the receiver in well-known fashion to recover the original input information.” Ex. 1007, 3:48-55.

242. Accordingly, Ozawa and Ordentlich disclose “a transmission circuit for transmitting the encoded wideband signal,” as required by claim 28.

243. Ordentlich also discloses a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal. Specifically, Ordentlich discloses that “the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. **The transmitted information is used at the receiver in well-known fashion to recover the original input information.**”

Ex. 1007, 3:48-55 (emphasis added); *see also id.* 5:40-66.

244. Accordingly, Ordentlich discloses “a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal,” as required by claim 28.

245. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

246. Accordingly, Ozawa in view of Ordentlich discloses all the limitations of claim 28.

f. Claim 29

247. Claim 29 of the '521 Patent recites:

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.

248. Claim 29 is similar to claim 2 except it depends from claim 28

249. Ozawa in view of Ordentlich discloses “[a] cellular mobile transmitter/receiver unit as defined in claim 28,” as shown above in ¶¶238-246.

250. Ozawa discloses “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 shown above in ¶¶105-106.

251. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

252. Accordingly, Ozawa in view of Ordentlich discloses all the limitations of claim 29.

g. Claim 32

253. Claim 32 of the '521 Patent recites:

32. A cellular mobile transmitter/receiver unit as defined in claim 28, wherein each pitch prediction error calculating device comprises:
a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;

- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- and d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

254. Claim 32 is similar to claim 5 except that it depends from claim 28.

255. Ozawa in view of Ordentlich discloses “[a] cellular mobile transmitter/receiver unit as defined in claim 28,” as shown above in ¶¶238-246.

256. Ozawa discloses “wherein each pitch prediction error calculating device comprises: a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector,” as shown above in ¶¶109-111.

257. Ozawa discloses “a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector,” as shown above in ¶¶112-113.

258. Ozawa discloses “an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector,” as shown above in ¶¶114-115.

259. Ozawa discloses “a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error,” as shown above in ¶¶116-119.

260. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

261. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 32.

h. Claim 35

262. Claim 35 of the '521 Patent recites:

35. A cellular mobile transmitter/receiver unit as defined in claim 28, 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.

263. Claim 35 is similar to claim 8 except that it depends from claim 28.

264. Ordentlich discloses “[a] cellular mobile transmitter/receiver unit as defined in claim 28,” as shown above in ¶¶238-246.

265. Ozawa discloses “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” as shown above in ¶¶133-137.

266. Ozawa discloses “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,” as shown above in ¶¶138-139.

267. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

268. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 35.

i. Claim 37

269. Claim 37 of the '521 Patent recites:

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.

270. Ozawa in view of Ordentlich discloses “[a] A network element, comprising: a transmitter including an encoder for encoding a wideband signal as recited in claim 10,” as shown above in ¶¶187-214.

271. Ordentlich further discloses a network element, stating that “[o]ne area of such improvements have come to be called code excited linear predictive (CELP) coders . . . Such techniques have found application, e.g., in voice grade **telephone channels**, including **mobile telephone channels**.” Ex. 1007, 1:25-40 (emphases added); *see also id.* 5:40-66.

272. Ordentlich also discloses a transmission circuit for transmitting an encoded wideband signal. Ordentlich states that “[s]hown are **the transmitter portion** at the top of the figure, the receiver portion at the bottom and the various parameters (j, g, M, β and A) that are transmitted via a communication channel 50. “ Ex. 1007, 2:40-43 (emphasis added). The transmitter portion is understood by one of skill in the art as a transmission circuit.

273. Ordentlich discloses transmitting a wideband signal. Specifically, Ordentlich states that “[i]n general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. The transmitted information is used at the receiver in well-known fashion to recover the original input information.” Ex. 1007, 3:48-55.

274. Accordingly, Ordentlich discloses “a transmission circuit for transmitting the encoded wideband signal,” as required by claim 37.

275. It would have been obvious to combine Ordentlich with Ozawa as shown above in ¶¶210-213.

276. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 37.

j. Claim 38

277. Claim 38 of the '521 Patent recites:

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.

278. Claim 38 is similar to claim 2 except that it depends from claim 37.

279. Ozawa in view of Ordentlich discloses “[a] network element as defined in claim 37,” as shown above in ¶271.

280. Ozawa discloses “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 shown above in ¶¶105-106.

281. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

282. Accordingly, Ozawa in view of Ordentlich discloses all the limitations of claim 38.

k. Claim 41

283. Claim 41 of the '521 Patent recites:

41. A network element as defined in claim 37, wherein each pitch prediction error calculating device comprises:

- a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;
- b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;
- c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;
- d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.

284. Claim 41 is similar to claim 5 except that it depends from claim 37.

285. Ozawa in view of Ordentlich discloses “[a] network element as defined in claim 37,” as shown above in ¶271.

286. Ozawa discloses “wherein each pitch prediction error calculating device comprises: a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector,” as shown above in ¶¶109-111.

287. Ozawa discloses “a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector,” as shown above in ¶¶112-113.

288. Ozawa discloses “an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector,” as shown above in ¶¶114-115.

289. Ozawa discloses “a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error,” as shown above in ¶¶116-119.

290. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

291. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 41.

I. Claim 44

292. Claim 44 of the '521 Patent recites:

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.

293. Claim 44 is similar to claim 8 except that it depends from claim 37.

294. Ordentlich discloses “[a] network element as defined in claim 37,” as shown above in ¶¶271.

295. Ozawa discloses “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” as shown above in ¶¶133-137.

296. Ozawa discloses “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,” as shown above in ¶¶138-139.

297. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

298. Accordingly, Ozawa in view of Ordentlich discloses all of the limitations of claim 44.

2. Claims 15, 33, And 42 Are Obvious Over Ozawa In View Of Ordentlich And G.729

a. Claim 15

299. Claim 15 of the '521 Patent recites:

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:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

300. Ozawa in view of Ordentlich discloses “[a]n encoder as defined in claim 14,” as shown above in ¶¶223-229.

301. G.729 discloses “wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector,” as shown above in ¶¶171-175.

302. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

303. It would have been obvious to one of skill in the art to combine Ozawa with G.729 as shown above in ¶¶176-178.

304. The reasons set forth above also apply to the combination of Ozawa and Ordentlich because this combination does not impact the way pitch gain is calculated. Moreover, Ordentlich also requires computation of a gain for scaling excitation vectors during the optimization process. *See* Ex. 1007, 3:31-32; Fig. 1. Thus, a person of ordinary skill in the art implementing the gain calculator 23 and the gain codebook 24 of Ozawa would have used established CELP principles to calculate the values of β'_k as the gain of k -th adaptive code vector of Ozawa. A person of ordinary skill in the art would also have known that the established principle for calculating the gain of adaptive code vector in the context of CELP

codec is through Equation 43 as codified in G.729. *Compare* Ex. 1004 (Equation 43) with Ex. 1008, 8:20-24 (U.S. 5,444,816 to Adoul et al., Equation 7); Ex. 1028, 9:25-43 (U.S. Patent No. 5,293,449 to Tzeng, Equation 9); Ex. 1029, 16:56-68 (U.S. Pat. No. 4,868,867 to Davidson, Equation 10). Thus, the '521 Patent simply claims a known mathematical formula for calculating the gain of a codevector.

305. Accordingly, Ozawa in view of G.729 and Ordentlich discloses all of the limitations of claim 15.

b. Claim 33

306. Claim 33 of the '521 Patent recites:

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:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

307. Ozawa in view of Ordentlich discloses “[a] cellular mobile transmitter/receiver unit as defined in claim 32,” as shown above in ¶¶255-261.

308. G.729 discloses “wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector,” as shown above in ¶¶171-175.

309. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

310. It would have been obvious to one of skill in the art to combine Ozawa with G.729 as shown above in ¶¶176-178.

311. It would have been obvious to one of skill in the art to combine Ozawa with G.729 and Ordentlich as shown above in ¶304.

312. Accordingly, Ozawa in view of G.729 and Ordentlich discloses all of the limitations of claim 33.

c. Claim 42

313. Claim 42 of the '521 Patent recites:

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:

$$b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$$

where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.

314. Ozawa in view of Ordentlich discloses “[a] network element as defined in claim 41,” as shown above in ¶¶285-291.

315. G.729 discloses “wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch

search target vector and $y^{(i)}$ is said convolved pitch codevector,” as shown above in ¶¶171-175.

316. It would have been obvious to one of skill in the art to combine Ozawa with Ordentlich as shown above in ¶¶210-213.

317. It would have been obvious to one of skill in the art to combine Ozawa with G.729 as shown above in ¶¶176-178.

318. It would have been obvious to one of skill in the art to combine Ozawa with G.729 and Ordentlich as shown above in ¶¶304.

319. Accordingly, Ozawa in view of G.729 and Ordentlich discloses all of the limitations of claim 42.

VI. SUPPLEMENTATION

320. This declaration, along with the attached appendices, is based on my present assessment of material and information currently available to me.

321. I hereby declare that all statements made herein of my own knowledge are true and that all statements made herein on information and belief are believed to be true. Further, these statements were made with the knowledge that willful false statements and the like so made are punishable by fine, imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of the above-identified patent.

Dated: March 10, 2016

Respectfully submitted,

By: Michael T. Johnson
Michael T. Johnson, Ph.D.

APPENDIX A

Curriculum Vitae

Michael T. Johnson

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Director of the Marquette Speech and Signal Processing Laboratory
P.O. 1881, Milwaukee, WI 53201-1881

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CURRICULUM VITAE EXECUTIVE SUMMARY

Publications: **36 journal papers, over 100 refereed publications/presentations**
 Citations >1300, h-index 19, i10-index 38 (per Google scholar)

Grant funding: **5 major grants, over \$2.5 million in total external funding**
 Multi-institution, interdisciplinary, international collaborations

Leadership roles: **Chair of MU Board of Graduate Studies (current)**
 Director of Speech and Signal Processing Lab (current)
 Director of Graduate Studies, EECE (2009-2012)
 Senior Visiting Professor, Tsinghua University (2008-2009, 2014-2015)
 Engineering Manager (1993-1996)

Honors: **Featured in “Groundbreaking Thinkers in Wisconsin” series 2007**
 Marquette COE Researcher of the Year 2006-2007
 Engineers and Scientists of Milwaukee Young Engineer of the Year 2006
 HKN EECE Teacher of the year 2005, 2014

RESEARCH SPECIALIZATION

- Speech Processing: speech recognition and enhancement, natural language processing
- Signal Processing: bioacoustics, microphone array processing, signal enhancement
- Pattern Recognition and Machine Learning: statistical estimation and classification

EDUCATION

Doctor of Philosophy, School of Electrical and Computer Engineering, August 2000

Purdue University, West Lafayette, IN GPA: 4.00/4.00

"Incorporating Prosodic Information and Language Structure into Speech Recognition Systems"

Advisor: Dr. Leah H. Jamieson

Master of Science in Electrical Engineering (concentration Digital Signal Processing), Dec. 1994

University of Texas at San Antonio, San Antonio, TX GPA: 4.00/4.00

"An Adaptive Texture Model Based on Markov Random Fields"

Advisor: Dr. Mita Desai

Bachelor of Science, Engineering with Electrical Concentration, April 1990

LeTourneau University, Longview, TX GPA: 3.60/4.00

Bachelor of Science, Computer Science and Engineering with minor in Mathematics, April 1989

LeTourneau University, Longview, TX GPA: 3.57/4.00

ACADEMIC WORK EXPERIENCE

8/2000 - present Marquette University, Milwaukee, WI

- *Full Professor, Electrical and Computer Engineering:* (2013 - present)
- *Director of Graduate Studies, Electrical and Computer Engineering:* (2009 - 2012)
- *Associate Professor with Tenure, Electrical and Computer Engineering* (2007 – present)
- *Assistant Professor, Electrical and Computer Engineering:* (2000 – 2007)

8/2008 – 8/2009 Tsinghua University, Beijing, China

- *Senior Visiting Faculty, Electronic Engineering Dept.:* While on sabbatical, taught several courses and collaborated on research in speech processing with Prof. Liu Jia.

9/1996 – 8/2000 Purdue University, West Lafayette, IN

- *Graduate Research Assistant, Audiology Dept.:* (5/97 to 5/00) Research focused on studying the consistency of speech articulator movement in children and young adults.
- *Graduate Research Assistant, ECE Dept.:* (1/97 to 1/98) Part of an NSF-funded research grant working on interfacing speech recognition and language processing systems.
- *Teaching Assistant, ECE Dept.:* (9/96 to 12/96 and 1/98 to 5/98) Assisted with a course in Speech Processing and taught a laboratory course in Digital Circuits.

1/1993 - 9/1993 University of Texas at San Antonio, San Antonio, TX

- *Graduate Research Assistant, UTSA Image Processing Lab:* Did research in the field of biological image processing as part of masters thesis in mammogram image analysis.

INDUSTRIAL WORK EXPERIENCE

9/1993 - 7/1996 SNC Manufacturing, Oshkosh, WI

- *Engineering Manager, Telecommunications Division* (11/94 - 7/96): Responsible for all new and existing product lines, including design, project management, and personnel.
- *Senior Engineer* (9/93-11/94): Project lead for new products, primarily in the area of SNC's Lyte Lynx line of fiber-optic interfaces for high-voltage environments.

9/1991 - 1/1993 Datapoint Corporation , San Antonio, TX

- *Engineer II:* Hardware engineer on the ArcnetPlus networking and MINX videoconferencing projects. Lead engineer on the PC HUB project for ArcnetPlus.

5/1990 - 9/1991 Micronyx, Inc. / Micro Technology Services, Inc., Richardson, TX

- *Design Engineer:* Extensive telecommunications and embedded microprocessor design, at both hardware and software levels for Micronyx' Services division and its spin-off company MTSI. Major projects included redesign of test consoles for telephone PBX systems and development of a hotel pay-per-view video system.

5/1989 - 8/1989 GTE , Ft. Wayne, IN *Engineering Intern:* Designed circuits for Business Services, including a project restructuring an analog phone system into a digital carrier system.

6/1988 - 8/1988 NBS , Ft. Wayne, IN *Electrical Technician:* Gained experience with troubleshooting circuit boards, soldering, and designing and building test equipment.

PUBLICATIONS

Peer-reviewed Journal Publications

1. Wei-Qiang Zhang, Cong Guo, Qiao Zhang, Jian Kang, Liang He, Jia Liu, and Michael T. Johnson, A Speech Enhancement Algorithm Based on Computational Auditory Scene Analysis. *Journal of Tianjin University*, in press, 2015.
(Chinese journal. Original language citation: 张卫强, 郭聪, 张乔, 康健, 何亮, 刘加, and Michael T. Johnson, 种基于计算听觉场景分析的语音增强算法, 天津大学学报.)
2. PM Scheifele, MT Johnson, M Fry, B Hamel, K Laclede, Vocal classification of vocalizations of a pair of Asian Small-Clawed otters to determine stress, *The Journal of the Acoustical Society of America*, 138 (1), EL105-EL109, 2015.
3. Liu Wei-Wei, Cai Meng, Zhang Wei-Qiang Zhang, Liu Jia, Johnson Michael T., “Discriminative Boosting Algorithm for Diversified Front-End Phonotactic Language Recognition,” *Journal of Signal Processing Systems*, May 2015.
4. Trawicki, Marek B. and Johnson, Michael T., “Beta-order minimum mean-square error multichannel spectral amplitude estimation for speech enhancement”, *International Journal of Adaptive Control and Signal Processing*, January 2015.
5. Arik Kershenbaum, Daniel Blumstein, Marie Roch, Michael T. Johnson, et. al., Acoustic sequences in non-human animals: a tutorial review and prospectus, *Biological Reviews*, 2014.
6. C Yu, KK Wójcicki, PC Loizou, JHL Hansen, MT Johnson, “Evaluation of the importance of time-frequency contributions to speech intelligibility in noise”, *The Journal of the Acoustical Society of America*, vol. 135, no. 5, May 2014, 3007-3016.
7. Marek B. Trawicki, Michael T. Johnson , “Speech enhancement using Bayesian estimators of the perceptually-motivated short-time spectral amplitude (STSA) with Chi speech priors”, *Speech Communication*, vol. 57, no. 2, February 2014, pp101-103.
8. Liu Weiwei, Zhang Weiqiang, Johnson Michael T., Liu Jia, “Homogenous ensemble phonotactic language recognition based on SVM supervector reconstruction”, *EURASIP Journal on Audio, Speech, and Music Processing* vol. 2014 no. 1, January 2014, pp 1-13.
9. Junhong Zhao, Wei-Qiang Zhang, Hua Yuan, Michael T Johnson, Jia Liu, Shanhong Xia, “Exploiting contextual information for prosodic event detection using auto-context”, *EURASIP Journal on Audio, Speech, and Music Processing*, vol. 2013, no. 1, December 2013 pp 1-14.
10. Marek B. Trawicki, Michael T. Johnson , “Distributed multichannel speech enhancement based on perceptually-motivated Bayesian estimators of the spectral amplitude”, *IET Signal Processing*, vol. 7, no.4, April 2013, pp. 337-344.
11. An Ji, Michael T. Johnson, Edward J. Walsh, JoAnn McGee, Doug L. Armstrong, Discrimination of individual tigers (*Panthera tigris*) from long distance roars, *The Journal of the Acoustical Society of America*, vol. 133 no. 3, March 2013, pp1762-1769.
12. Yongzhe Shi, Weiqiang Zhang, Jia Liu, Michael T. Johnson, “RNN language model with word clustering and class-based output layer”, *EURASIP Journal on Audio, Speech, and Music Processing* vol. 2013 no. 1, January 2013, pp1-7.
13. Peter M. Scheifele, Michael T. Johnson, David C. Byrne, John G. Clark, Ashley Vandlik, Laura W. Kretschmer, Kristine E. Sonstrom, “Noise impacts from professional dog grooming forced-air dryers”, *Noise and Health*, vol. 14 no. 60, October 2012, p224-226.

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Peer-reviewed Conference Publications and Presentations

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38. Yi Liu, Yao Tian, Liang He, Jia Liu, Michael T Johnson, Simultaneous Utilization of Spectral Magnitude and Phase Information to Extract Supervectors for Speaker Verification, ASV Spoofing Challenge 2015, presented at Interspeech 2015, September 2015.
39. Gui Jin, Michael T Johnson, Jia Liu, Xiaokang Lin, Voice conversion based on Gaussian mixture modules with Minimum Distance Spectral Mapping, 2015 5th International Conference on Information Science and Technology (ICIST), April 2015, pp. 356-359.

40. Berry, Jeffrey, Cassandra North, and Michael T. Johnson, Dynamic aspects of articulating with a virtual vocal tract in dysarthria, International Conference on Motor Speech, 2014.
41. Berry, Jeffrey, A Kolb, C North, and Michael T. Johnson, Acoustic and Kinematic Characteristics of Vowel Production through a Virtual Vocal Tract in Dysarthria, Interspeech, Singapore, September 2014.
42. Berry, Jeffrey, John Jaeger, M Wiedenhoeft, B Bernal, and Michael T. Johnson, Consonant Context Effects on Vowel Sensorimotor Adaptation, Interspeech, Singapore, September 2014.
43. Ji, An, Michael T. Johnson, and Jeffrey Berry, Palate-referenced Articulatory Features for Acoustic-to-Articulator Inversion, Interspeech, Singapore, September 2014.
44. Jianglin Wang and Michael T. Johnson, "Physiologically-motivated Feature Extraction for Speaker Identification", International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2014, Florence, Italy.
45. Jeffrey Berry, Cassandra North, and Michael T. Johnson, "Sensorimotor adaptation of speech using real-time articulatory resynthesis", International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2014, Florence, Italy.
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47. Jeffrey Berry, An Ji, and Michael T. Johnson, "Dynamic aspects of vowel production in Mandarin-accented English", American Speech-Language-Hearing Association 2013, Chicago, IL, November 2013.
48. Jeffrey Berry, Christine Belchel, Cassandra North, and Michael T. Johnson, "Learning novel articulatory-acoustic mappings in dysarthria", American Speech Language Hearing Association 2013, Chicago, IL, November 2013.
49. Jianglin Wang and Michael T. Johnson, "Vocal source features for bilingual speaker identification", China SIP, July 2013, Beijing China.
50. An Ji, Michael T. Johnson, Jeffrey Berry, "Articulatory space calibration in 3D Electromagnetic Articulography", China SIP, July 2013, Beijing China.
51. Jeffrey Berry, Cassandra North, Benjamin Meyers, Michael T. Johnson, Speech sensorimotor learning through a virtual vocal tract, Proceedings of Spring 2013 Meetings on Acoustics, Montreal, June 2013.
52. An Ji, Jeffrey Berry, Michael T. Johnson, Vowel production in Mandarin accented English and American English: Kinematic and acoustic data from the Marquette University Mandarin accented English corpus, Proceedings of Spring 2013 Meetings on Acoustics, Montreal, June 2013.
53. Jianglin Wang, MT Johnson, "Residual Phase Cepstrum Coefficients with Application to Cross-lingual Speaker Verification", Interspeech 2012, Portland, September 2012.
54. Trawicki, Marek B. and Michael T. Johnson. "Improvements of the Beta-Order Minimum Mean-Square Error (MMSE) Spectral Amplitude Estimator using Chi Priors," Interspeech 2012, Portland, September 2012.

55. An Ji, Michael T. Johnson, Jeffrey Berry, "Tracking articulator movements using orientation measurements", 2012 International Conference on Audio, Language and Image Processing (ICALIP), Shanghai, China, July 2012.
56. Marek B. Trawicki, Michael T. Johnson, An Ji, Tomasz S. Osiejuk, "Multichannel speech recognition using distributed microphone signal fusion strategies", 2012 International Conference on Audio, Language and Image Processing (ICALIP), Shanghai, China, July 2012.
57. Jianglin Wang, An Ji, Michael T. Johnson, "Features for phoneme independent speaker identification", 2012 International Conference on Audio, Language and Image Processing (ICALIP), Shanghai, China, July 2012.
58. Michael T. Johnson and Patrick Clemins, "Individual Identification using Hidden Markov Models for Population Monitoring and Assessment", Invited Presentation to Special Session "Topical Meeting on Signal Processing of Subtle and Complex Acoustic Signals in Animal Communication", Acoustical Society of America Spring 2010 Meeting, Baltimore, Maryland, April 2010.
59. Marek Trawicki and Michael T. Johnson, "Optimal Distributed Microphone Phase Estimation," International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2009, Taiwan, April 2009, pp 2177-2180.
60. Yao Ren and Michael T. Johnson, "Auditory Coding based Speech Enhancement", International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2009, Taiwan, April 2009, pp 4685-4688.
61. Marek B. Trawicki, Yao Ren, Michael T. Johnson, Tomasz S. Osiejuk, Distributed Multi-Channel Speech Enhancement of Ortolan Bunting (*Emberiza Hortulana*) Vocalizations, Acoustic Communication by Animals, Corvallis, Oregon, 2008, pp. 276-277.
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76. Anthony D. Ricke, Richard J. Povinelli, Michael T. Johnson. "Segmenting Heart Sound Signals," Computers in Cardiology, Leon, France, September 2005.
77. Marek Trawicki and Michael T. Johnson, "Automatic Song-Type Classification and Speaker Identification of Norwegian Ortolan Bunting (*Emberiza Hortulana*) Vocalizations", IEEE International Conference on Machine Learning in Signal Processing (MLSP), Mystic, Connecticut, September 2005.
78. Patrick J. Clemins and Michael T. Johnson, "Unsupervised Classification of Beluga Whale Vocalizations", Spring 2005 Meetings of the Acoustical Society of America, Vancouver, May 2005.
79. Kevin M. Indrebo, Richard J. Povinelli, Michael T. Johnson. "Third-Order Moments of Filtered Speech Signals For Robust Speech Recognition," International Conference on Non-Linear Speech Processing (NOLISP) 2005, Barcelona, Spain, 151-157.
80. Jidong Tao and Michael T. Johnson, "Comparison and Evaluation of Animal Vocalization Enhancement Techniques", Fall 2004 Meetings of the Acoustical Society of America, San Diego, November 2004.

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82. Heather E. Ewalt and Michael T. Johnson, "Combining Multi-source Wiener Filtering with Parallel Beamformers to Reduce Noise from Interfering Talkers", International Conference on Signal Processing (ICSP) 2004, Beijing, August 2004.
83. Kevin M. Indrebo, Richard J. Povinelli, and Michael T. Johnson, "A comparison of reconstructed phase spaces and cepstral coefficients for multi-band phoneme classification", ICSP 2004, Beijing, August 2004.
84. Patrick Clemins and Michael T. Johnson, "Generalized perceptual features for animal vocalization classification", Spring 2004 Meetings of the Acoustical Society of America, New York, May 2004.
85. Andrew Lindgren, Michael T. Johnson, and Richard J. Povinelli, "Joint Frequency Domain and Reconstructed Phase Space Features for Speech Recognition", International Conference on Acoustics, Speech and Signal Processing (ICASSP) 2004, Montreal, May 2004.
86. Mike Zimmerman, Richard J. Povinelli, Michael T. Johnson, and Kris Ropella, "A Reconstructed Phase Space Approach for Distinguishing Ischemic from Non-Ischemic ST Changes using Holter ECG Data", Computers in Cardiology, Thessalonica, Greece, September, 2003.
87. Patrick Clemins and Michael T. Johnson, "Automatic Classification of African Elephant (*Loxodonta africana*) Follicular and Luteal Rumbles", 1st International Conference on Acoustic Communication by Animals, Baltimore, Maryland, July 2003, pp 81-82.
88. Patrick Clemins and Michael T. Johnson, "Automatic Type Classification and Speaker Identification of African Elephant (*Loxodonta Africana*) Vocalizations", Spring 2003 Meetings of the Acoustical Society of America, Nashville, May 2003.
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91. Jinjin Ye, Michael T. Johnson, and Richard J. Povinelli, "Study of attractor variation in the reconstructed phase space of speech signals, NOLISP 2003, Le Croisic, France, May 2003.
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94. Andrew C. Lindgren, Michael T. Johnson, and Richard J. Povinelli, "Speech Recognition using Reconstructed Phase Space Features", Proceedings of the International Conference on Acoustics, Speech and Signal Processing (ICASSP) 2003, Hong Kong, April 2003, pp I-60 – I-63.

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97. Xiaolin Liu, Richard Povinelli, and Michael T. Johnson, "Detecting Determinism in Speech Phonemes", IEEE Digital Signal Processing Workshop 2002, October 2002.
98. Jinjin Ye, Richard J. Povinelli, and Michael T. Johnson, "Phoneme Classification Using naïve Bayes Classifier in Reconstructed Phase Space", IEEE Digital Signal Processing Workshop 2002, October 2002.
99. Richard J. Povinelli, Michael T. Johnson, Nabeel A.O. Demerdash, and John F. Bangura, "A Comparison of Phase Space Reconstruction and Spectral Coherence Approaches for Diagnostics of Bar and End-Ring Connector Breakage and Eccentricity Faults in Polyphase Induction Motors using Motor Design Particulars", IEEE Industry Applications Society meetings 2002, October 2002.
100. Richard J. Povinelli, Felice M. Roberts, Kristina M. Ropella, and Michael T. Johnson, "Are Nonlinear Ventricular Arrhythmia Characteristics Lost, As Signal Duration Decreases?", Computers in Cardiology 2002, September 2002.
101. Patrick Clemins and Michael T. Johnson, "Automatic Speech Recognition and Speaker Identification of Animal Vocalizations", Measuring Behavior 2002, August 2002, pp 41-46.
102. Patrick Clemins, Heather Ewalt and Michael Johnson, "Time-aligned SVD Analysis for Speaker Identification", International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 2002, May 2002, p 4160.
103. Michael T. Johnson and Leah H. Jamieson, "Temporal Features for Broadcast News Segmentation", ISCA Workshop on Prosody in Speech Understanding and Recognition, October 2001.
104. Michael T. Johnson and Mary P. Harper, "Near Minimal Weighted Word Graphs for Post-processing Speech", International Workshop on Automatic speech Recognition and Understanding (ASRU), December 1999.
105. M.P. Harper, M.T. Johnson, L.H. Jamieson, S.A. Hockema, and C.M. White, "Interfacing a CDG Parser with an HMM Word Recognizer Using Word Graphs", Proceedings of the International Conference on Acoustics, Speech, and Signal Processing (ICASSP) 1999, March 1999, pp 733-736.
106. Michael T. Johnson, Leah H. Jamieson, and Mary P. Harper "Interfacing Acoustic Models with Natural Language Processing Systems", Proceedings of the International Conference on Spoken Language Processing (ICSLP) 1998, December 1998, pp 2419-2422.
107. A.M. Supranant, S.L. Hura, M.P. Harper, L.H. Jamieson, G. Long, S.M. Thede, A. Rout, T.H. Hsueh, S.A. Hockema, M.T. Johnson, J.B. Laflen, P. Srinivasa, and C.M. White, "Familiarity and Pronouncibility of Nouns and Names: The Purdue Proper Name Database", Spring 1998 Meetings of the Acoustical Society of America, June 1998.

108. Michael Johnson and Mita Desai, "An Adaptive Approach for Texture Modelling", Proceedings of the International Conference on Image Processing (ICIP) 1994, November 1994.
109. Harold Longbotham, Michael Johnson, Jack Harris, and Redouan Rouzky, "The FatBear: a nonarithmetic pico filter", Proceedings of the SPIE, Image Algebra and Morphological Image Processing IV, July 1993, pp 128-139.

Selected Invited Presentations

- Invited speaker, Shanghai Jiaotong University Colloquium Series, "Electromagnetic Articulography for Speaker Independent Acoustic-to-Articulatory Inversion, April 2015.
- Invited speaker, University of Memphis Institute of Intelligent Systems Colloquium Series, "Applying Speech Technology to Animal Vocalizations", April 2014.
- Invited speaker, NIMBioS Investigative Workshop on Analyzing Animal Vocal Sequences, "Using HMMs to model vocal structure and sequence for classification and identification", October 2013.
- Featured Keynote Speaker, GE Healthcare Technology symposium, "Innovations and Applications of Artificial Intelligence and Bayesian Modeling", August 2011.
- "Individual Identification using Hidden Markov Models for Population Monitoring and Assessment", Invited Presentation to Special Session "Topical Meeting on Signal Processing of Subtle and Complex Acoustic Signals in Animal Communication", at Acoustical Society of America Spring 2010 Meeting, Baltimore, Maryland, April 2010.
- Featured speaker at ShARE World Seminar 2008, "Innovation in Computation", Tsinghua University, Beijing China, December 2008.
- Keynote speaker for annual Tau Beta Pi banquet, "The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations," November 2007.
- Presentation to Beijing Institute of Technology annual colloquia, "The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations," Beijing, China, May 2007.
- Presentation to Tsinghua University Center for Speech Technology, "The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations," Beijing, China, May 2007.
- Presentation to the Chinese Academy of Sciences, "The Dr. Dolittle Project: Applying Speech Processing to Animal Vocalizations," Institute of Acoustics, Beijing, China, May 2007.
- Michael T. Johnson, "The Dr. Dolittle Project: Applying speech technology to animal vocalizations", Sigma Xi annual banquet, April 27, 2007.
- Michael T. Johnson, "Speech Research at Marquette – The Dr. Dolittle Project", Marquette Trustees Committee on Academic Affairs and Planning, March 3, 2004.
- Michael T. Johnson, "Speech Recognition and Chaos Theory: A new approach to signal analysis", Lynde Bradley Science Club, Rockwell Automation, November 11, 2003.

- Michael T. Johnson, “The Dr. Dolittle Project: Applying human speech processing algorithms to other species”, UW Milwaukee Computer Science Department Colloquium, November 7, 2003.
- Michael T. Johnson, “Talking with the Animals: Automatic Classification and Identification of Animal Vocalizations using Speech Technology”, Keynote Speaker for Physics Club of Milwaukee Annual Banquet, June 5, 2002.
- Michael T. Johnson, “Speech Processing Research at Marquette (Elephants and Cocktail Parties)”, COE National Research Council meeting, April 26, 2002.
- Michael T. Johnson, “Speech and Signal Processing Research at Marquette”, Invited speaker at Sigma Xi annual meeting, February 20, 2002.

PhD Dissertation and MS Thesis

- Michael T. Johnson, "Incorporating Prosodic Information and Language Structure into Speech Recognition Systems", PhD Dissertation, Purdue University School of Electrical and Computer Engineering, August 2000.
- Michael T. Johnson, "An Adaptive Texture Model Based on Markov Random Fields", Masters Thesis, University of Texas at San Antonio, August 1994.

Student PhD Dissertations (primary advisor)

- An Ji, “Speaker Independent Acoustic-to-Articulatory Inversion”, PhD Dissertation, Marquette University, December 2014.
- Jianglin Wang, "Physiologically-motivated feature extraction methods for speaker recognition", PhD Dissertation, Marquette University, December 2013.
- Jidong Tao, “Acoustic Model Adaptation for Automatic Speech Recognition and Animal Vocalization Classification”, PhD Dissertation, Marquette University, May 2009.
- Marek Trawicki, “Distributed Multichannel Processing for Signal Enhancement”, PhD Dissertation, Marquette University, May 2009.
- Kuntoro Adi, “Hidden Markov model based animal acoustic censusing: learning from speech processing technology”, Ph.D. Dissertation, Marquette University, May 2008.
- Patrick J. Clemins, “Automatic Classification of Animal Vocalizations”, PhD Thesis, Marquette University, May 2005.

Student MS Theses (primary advisor)

- Andrew Kolb, Development of Software Tools and Analysis methods for the Use of Electromagnetic Articulography Data in Speech Research”, Masters Thesis, Marquette University, May 2015.
- Zac Slavens, “Generalized interpolation applied to MR image magnification and gradient nonlinearity correction”, Masters Thesis, Marquette University, May 2008.
- Xi Li, “SPeECH Feature Toolbox (SPEFT) Design, and Emotional Speech Feature Extraction”, M.S. Thesis, Marquette University, August 2007.

- Anthony D. Ricke, “Automatic Frame Length, Frame Overlap, and Hidden Markov Model Topology for Automatic Speech Recognition of Animal Vocalizations”, M.S. Thesis, Marquette University, December 2006.
- Jinjin Ye, “Speech Recognition Using Time Domain Features from Phase Space Reconstructions”, M.S. Thesis, Marquette University, May 2004.
- Franck Hounkpevi, “Triphone Creation Through Rule-based Trajectory Interpolation for Continuous Speech Recognition”, M.S. Thesis, Marquette University, May 2003.
- Xiaolong Yuan, “Auditory Model-Based Bionic Wavelet Transform for Speech Enhancement”, M.S. Thesis, Marquette University, May 2003.
- Andrew Lindgren, “Speech Recognition using Features extracted from Phase Space Reconstructions”, M.S. Thesis, Marquette University, May 2003.
- Heather Ewalt, “Speech Signal Enhancement Using a Microphone Array”, M.S. Thesis, Marquette University, December 2002.

Selected News and Magazine Articles

- “Listening to the Animal Kingdom”, Discover Magazine, Marquette University, 2008, pp 14-15.
- “Decoding Calls of the Wild”, by Mark Johnson, feature article in Milwaukee Journal Sentinel, part of “Brainpower: Groundbreaking Thinkers in Wisconsin”, October 2008.
- “Say What?”, cover article on Dr. Dolittle Project, Weekly Reader Science, Spring 2007.
- “Animal Talk”, by Elena Cabral, Scholastic News Edition 5/6, December 18, 2006.
- “Animal “Speech” Project Aims to Decode Critter Communication”, by Maryann Mott, National Geographic Online, <http://news.nationalgeographic.com/news/2006/09/060926-dolittle-project.html>, September 2006.
- “Parsing the Puffin’s Patois”, article on Dr. Dolittle Project by Rachel Metz, Wired News online, July 2006.
- “Quacking the Code”, by Katie Pelech, article in Milwaukee Magazine, March 2006, p. 30.
- “What Jumbo tells Dumbo” by Alex Antunes, feature article in Computers in Science and Engineering (CISE) magazine, published by IEEE Computer Society, September-October 2005, pp 3-6.
- “Discoveries with Disney”, article about collaborative project with Disney’s Animal Kingdom published in MU Magazine, Fall 2003, p. 17.

GRANTS

Major Grants Funded

1. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), “RI: Small: Speaker Independent Acoustic-Articulator Inversion for Pronunciation Assessment”, National Science Foundation CISE Directorate, 2013. Total Budget \$449,643 over 3 years.

2. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, National Science Foundation CISE Directorate, 2011. Total Budget: \$149,896.00 over 1 year.
3. Michael T. Johnson (PI), Elizabeth Mugganthalder (Fauna Communications Research Institute), Peter Scheifele (National Undersea Research Center, University of Connecticut), Mike Darre (Animal Sciences, University of Connecticut), Anne Savage (Disney Animal Kingdom), “The Dr. Dolittle Project: A Framework for Classification and Understanding of Animal Vocalizations”, NSF IT-R program under CISE Directorate, 2003. Total Budget: \$1,200,000 over 4 years.
4. Michael T. Johnson (PI) and Richard J. Povinelli, “Integration of Stochastic and Dynamical Methods for Speech Technology”, NSF IT-R program under CISE directorate, 2001. Total Budget: \$360,000 over 3 years.
5. Richard J. Povinelli (PI), Nabeel A. O. Demerdash, Edwin Yaz, and Michael T. Johnson, “A Novel Approach to Fault Modeling, Diagnostics, and Prediction in Motor Drive Systems”, NSF Division of Controls, Networks, and Computational Intelligence, 2003. Total Budget: \$340,000 over 3 years.

Additional Grants Funded

1. Jeffrey Berry (PI) and Michael T. Johnson (co-PI), “Speech Rehabilitation Using a Virtual Vocal Tract”, Marquette Regular Research Grant (RRG). Funded \$6,000 in 2014-2015.
2. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), Research Experiences for Undergraduates (REU) supplement to “RI: Small: Speaker Independent Acoustic-Articulator Inversion for Pronunciation Assessment”, NSF REU program, CISE directorate. Funded \$6,000 in 2013-2014, \$6000 in 2014-2015, \$6000 in 2015-2016.
3. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), OISE International supplement to “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, co-funded by CISE and OISE directorates. Funded \$19,840 in 2012-2013.
4. Michael T. Johnson (PI) and Jeffrey Berry (co-PI), Research Experiences for Undergraduates (REU) supplement to “EAGER: Acoustic-Articulator Modeling for Pronunciation Analysis”, NSF REU program, CISE directorate. Funded \$8,000 in 2011-2012.
5. Richard J. Povinelli (PI) and Michael T. Johnson (PI), “An examination of phoneme confusability in spoken English”, Marquette Regular Research Grant (RRG). Funded \$15,000 in 2011-2012.
6. Michael T. Johnson (PI) et. al., Research Experiences for Undergraduates (REU) supplement to “The Dr. Dolittle Project”, NSF REU program, CISE directorate. Funded at \$6,000 in 2003-04, \$15,000 in 2004-2005, \$12,000 in 2005-2006, \$15,000 in 2006-2007 Total Budget: \$48,000.
7. Michael T. Johnson (PI) and Richard J. Povinelli, REU supplement to “Integration of Stochastic and Dynamical Methods for Speech Technology”, NSF REU program, CISE directorate. Funded at \$15,000 in 2001-02, 2002-03, and 2003-04, Total Budget: \$45,000.
8. Richard J. Povinelli (PI), Nabeel A. O. Demerdash, Edwin Yaz, and Michael T. Johnson, REU supplement to “A Novel Approach to Fault Modeling, Diagnostics, and Prediction in Motor Drive Systems”, NSF REU Program, CNCI Division. Funded at \$6,000 in 2003-04 and 2004-05. Total Budget to-date: \$6,000.

PROFESSIONAL ACTIVITIES

Technical Committees and Publication editing

- Area Editor, Speech Enhancement, Speech Communication Journal, 2014-present
- Member, IEEE Signal Processing Society Speech and Language Technical Committee (SLTC), 2015-present

Publication Review Participation

- Speech Communication Journal
- International Conference on Acoustics, Speech and Signal Processing
- Interspeech IEEE Automatic Speech and Recognition Workshop
- IEEE Transactions on Speech and Audio Processing
- Journal of the Acoustical Society of America
- IEEE Sensors Journal
- IEEE Transactions on VLSI
- IEEE Transactions on Signal Processing
- IEEE Signal Processing Letters
- IEEE Transactions on Neural Systems
- Journal of Applications of Signal Processing
- Animal Behavior Journal
- EURASIP Journal on Audio, Speech and Music Processing
- International Journal on Adaptive Control and Signal Processing
- Biosystems Engineering Journal
- Acoustics Research Letters Online
- International Conference on Decision and Control
- Midwest Symposium on Circuits and Systems
- Iasted Circuits and Systems Conference
- Workshop on Signal Processing and Applications
- American Controls Conference
- Journal of Zhejiang University Science C (Computers & Electronics)
- Journal of Medical Engineering & Physics
- Open Signal Processing Journal
- IEEE Spoken Language Technology workshop
- International Conference on Signal Processing (ICSP)
- Journal of Computers
- International Journal of Electronics and Communications
- Franklin Institute journal

Funding Review Participation

- National Science Foundation
Have reviewed for CCLI, CAREER, CONICYT, and ITR programs, as well as various regular programs in the CISE and Biological Sciences directorate
- International: Austrian Science Fund, Icelandic Research Fund
- Medical College of Wisconsin Clinical and Translational Science Institute (CTSI)

Professional and honorary society memberships

- Senior Member IEEE, membership 1994-present
- IEEE Signal Processing Society, 1998-present
- Acoustical Society of America (ASA), 2001-present
- Senior Member Sigma Xi, 2001-present
- Eta Kappa Nu, 1996-present
- Upsilon Pi Epsilon, 2001-present
- Association of Computer Machinery (ACM), 1998-2010
- IEEE Computer Society, 1998-2010
- IEEE Circuits and Systems Society, 2001-2009
- Association of Computational Linguistics (ACL), 1998-2002
- International Speech Communication Association (ISCA), 2001-2006

Other professional activities and awards

- Featured in “Groundbreaking Thinkers in Wisconsin” article series 2007
- Researcher of the Year, Marquette College of Engineering, 2006-2007
- Young Engineer of the Year, Engineers and Scientists of Milwaukee (ESM), 2006
- Registered Professional Engineer, State of Wisconsin

TEACHING AND EDUCATIONAL ACTIVITIES

Courses Taught

Undergraduate

EECE Freshman Seminar
Computer Hardware
Digital Logic Design
Linear Systems Analysis
Embedded Systems Design

Graduate and Joint Undergraduate/Graduate

Digital Signal Processing
Speech Processing
Optimal/Adaptive Signal Processing
Pattern Recognition
Analysis of Algorithms
Information Theory
Professional Research Writing

Teaching awards

Eta Kappa Nu (HKN) honor society EECE Teacher of the Year, 2014, 2005

Selected Teaching, Education, and Pedagogy activities

Coordinator for EECE Freshman Seminar
Participant in KEEN/Lafferty Teaching Development Workshop, Fall 2015
Mentored numerous Research Experience for Undergraduates (REU) through NSF grants
International Teaching Experience at Tsinghua University in China, 2008-2009, 2011, 2013, 2014-2015
Signals and Systems Concept Inventory site coordinator for NSF study, 2003-2006
Developed and gave numerous DSP/Audio Workshops for middle and high school outreach
Participated with IEEE Signal Processing Education Technical Committee activities
Developed introductory DSP module for General Engineering freshman course
Participated in MU Assessment seminar by Barbara Wolvoord, May 2005
Participated in NSF Engineering Education Scholars (EES) program for new faculty
Participated in Prentice Hall Symposium on Education, Chicago, IL, 2001
Participated in NSF/ASEE Visiting Scholars Teaching Workshop, Spring 2001
Participated in NSF/ASEE Visiting Scholars Teaching Workshop, Fall 2000
Member, Advisory committee MU Center on Teaching and Learning (2005-2010)

Senior Design Advising

- Rehabilitative Acoustic Synthesis System, 2013-2014
- Rockwell Asset Management System, 2010-2011
- GE SVO Test Automation Unit, 2007-2008
- Harley Davidson Calibration Test Meter, 2006-2007
- Disney's Animal Kingdom Mobile Discovery Station, 2003-2004
- Building Remote Motoring System, 2003-2004
- Voice Authentication Security System, 2002-2003
- Multiple Simulator Device Interfaces for Test Automation, 2002-2003
- Audio transcription database system, 2001-2002
- Voice authentication system, 2001-2002
- Airplane check-list voice recognition system, 2000-2001

SERVICE ACTIVITIES

University:

- University Board of Graduate Studies (2009-present; chair, 2012-2014 and 2015-present)
- University Search Committee, Vice Provost for Enrollment Management (2015-present)
- University Steering committee on academic integrity (2012-2014)
- Subcommittee on academic integrity (2011)
- COF, Subcommittee on Nominations and Elections (chair, 2004-2007).
- Advisory Committee for Center on Teaching and Learning (2005-2010)
- Faculty mentor for Dissertation Boot Camps (2008, 2010)

College:

- COE Research Activity Committee (2011)
- COE Dean Search Committee (2009)
- COE Freshman Programs Committee (2006-2007)
- COE Dean Search committee (2003)
- COE Discovery Towers Building Committee (2004-2005)

Interdepartmental:

- BME Biocomputer Program Advisory Board (2001-present)
- Civil and Environmental Faculty Search Committee (2011-2012)

Departmental:

- EECE Graduate Committee (2000-2003, 2005-present)
- EECE Undergraduate Committee (2003-2008, 2012-2015)
- EECE Director of Graduate Studies (2009-2012)
- EECE Department Program Assessment Coordinator (2009-2012)
- EECE Publicity Committee (2004-present)
- EECE Goals Committee (2000-2001)
- EECE library representative (2001-2005)
- Numerous EECE faculty search and other subcommittees

Selected service activities within Marquette University

- Faculty advisor for Marquette's HKN chapter and Sigma Phi Delta engineering fraternity
- Active participant in EECE and COE open houses since starting at MU in 2000
- Regular blackjack dealer for annual engineering student council casino nights
- Regular participant in MU parent lunch program
- Panelist for University Advancement workshop, 2011
- ORSP Brown Bag panelist "One thing led to another" research series, 2010
- Participated in MU Assessment seminar by Barbara Wolvoord, May 2005.
- Attended First Annual Institute for Faculty: Career and Time Management, January 2005
- COE representative for "We are Marquette" World Café Conversation, 2004
- ORSP Brown Bag panelist on Pre-proposal Contact with Sponsors, 2003
- Gave research presentation to COE National Advisory Council, 2002
- Numerous university, college, and department student and student group activities
- Numerous research talks at Marquette, including EECE colloquium, EECE freshman seminar, research seminars, and others

Selected service and outreach activities

- Guest speaker, MOTIF English corner, Beijing, China. “Diversity”, September 2014, “Humor”, March 2015, “Languages of the World”, June 2015.
- Senior Visiting Scholar, Tsinghua University, China, 2014-2015, 2008-2009, summer 2011, summer 2013
- COE iPalm2 outreach program to high school students, August 2011, 2012, 2013
- Represented MU and COE on visits to Harbin Institute of Technology, China, 2011 (Have traveled to China on several occasions representing Marquette)
- Spoke to students at Rosenwald Dunbar elementary school in Nicholasville, KY, about the Dolittle project and careers in engineering, 2008
- Spoke to students at Cardinal Valley elementary school in Lexington, KY, about the Dolittle project and careers in engineering, 2006
- Participated in Marquette University High School Junior Career Day, 2005
- Spoke to “Future Problem Solvers” gifted program at Templeton Middle School about Artificial Intelligence, 2005
- Participated in Innovation Mondays meetings with local industry, 2004-2005
- EECE representative and presenter for GEMS Advanced Education Fair, 2004
- Spoke at MSOE about graduate school opportunities in EECE, 2002-2003
- Presented at Lynde Bradley Science Club, Rockwell Automation, 2003
- Presented at UW Milwaukee Computer Science Department Colloquium, 2003
- Was keynote Speaker at Physics Club of Milwaukee annual banquet, 2002
- Spoke at UWM Elementary Education class on careers in science and technology, 2002
- Participated in Nathan Hale H.S. Career Day regarding careers in engineering, 2002
- Member, Eastbrook Church & worship choir
- Numerous other volunteer and community activities

APPENDIX B

List of Materials Considered
U.S. Patent No. 7,260,521
File history of U.S. Patent No. 7,260,521
ITU-T Recommendation G.729, "Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP)," March 1996
ITU-T Recommendation G.728, "Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction," September 1992
U.S. Patent No. 5,797,119 ("Ozawa")
U.S. Patent No. 5,235,669 ("Ordentlich")
U.S. Patent No. 5,444,816 ("Adoul")
Atal, B. and Remde, J., "A New Model of LPC Extension for Producing Natural-Sounding Speech at Low Bit Rates," <i>IEEE</i> , pp. 614-617 (1982)
Spanias, A., "Speech Coding: A Tutorial Review," Proceedings of the <i>IEEE</i> , 82(10):1539-82 (1994)
Federal Coding Standard 1016 (February 14, 1991)
Schroeder, M. and Atal, B., "Code-Excited Linear Prediction (CELP): High Quality Speech at Very Low Bit Rates," <i>IEEE</i> , pp. 937-940 (1985)
Honkanen, T., et al., "Enhanced Full Rate Speech Codec for IS-136 Digital Cellular System," <i>IEEE</i> , pp. 731-734 (1997)
ETSI – GSM Digital Cellular Telecommunications System; Enhanced Full Rate (EFR) Speech Transcoding (GSM 06.60) (1996)
ITU G.722.2 Series G: Transmission Systems and Media, Digital Systems and Networks (2002)
Gersho, A. and Cuperman, V., "Vector Quantization: A Pattern-Matching Technique for Speech Coding," <i>IEEE</i> , pp. 15-21 (1983)
Chan, W., et al., "Enhanced Multistage Vector Quantization by Joint Codebook Design," <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 40(11):1693-1697 (1992)
Singhai, S. and A. Bishnu, "Improving Performance of Multi-Pulse LPC Coders at Low Bit Rates," <i>IEEE</i> , pp. 1.3.1-1.3.4 (1984)
Ramachandran, R. and Kabal, R., "Pitch Prediction Filters in Speech Coding," <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 37(4):467-478 (1989)
Kleijn, W. and Ketchum, R., "Improved Speech Quality and Efficient Vector Quantization in SELP," <i>IEEE</i> , pp. 155-158 (1988)

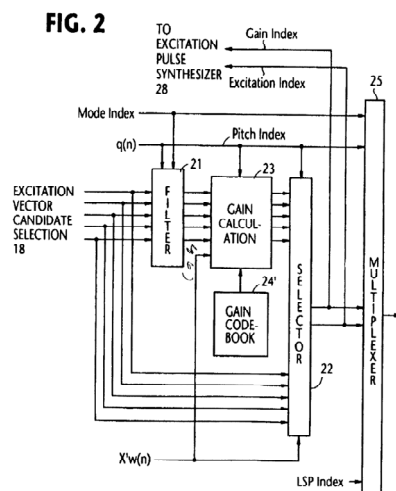
Marques, J, et al., “Improved Pitch Predication With Fractional Delays in CELP Coding,” <i>Filters in Speech Coding</i> , <i>IEEE</i> , 665-668 (1990)
Chen, J., and Gersho, A., “Adaptive Postfiltering for Quality Enhancement of Coded Speech,” <i>IEEE</i> , 3(1), pp. 59-71 (1995)
ITU G.722 – General Aspects of Digital Transmission Systems Terminal Equipments (1988)
Cheng, Y., et al., “Statistical Recovery of Wideband Speech From Narrowband Speech,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 2(4):544-548 (1994)
Ordentlich, E. and Shoham, Y., “Low-Delay Code-Excited Linear-Predictive Coding of Wideband Speech at 32 KBPS,” <i>IEEE</i> , pp. 9-12 (1991)
Avendano, C., et al., “Beyond Nyquist: Towards the Recovery of Broad-Bandwidth Speech from Narrow-Bandwidth Speech,” <i>Eurospeech</i> , 95 (1995)
J. Schnitzler, A 13.0 KBIT/S Wideband Speech Codec Based on SB-ACELP, <i>IEEE</i> , pp. 157-161 (1998)
U.S. Patent 5,293,449 (“Tzeng”)
U.S. Patent 4,868,867 (“Davidson”)
Kroon, P., “Regular-Pulse Excitation – A Novel Approach to Effective and Efficient Multipulse Coding of Speech,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 34(5):1054-1063 (1986)
Atal, B. and Schroeder, M., “Predictive Coding of Speech Signals and Subjective Error Coding,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 27(3):247-254 (1979)
Rabiner, L., “On the Use of Autocorrelation Analysis for Pitch Detection,” <i>IEEE Transactions on Acoustics, Speech, and Signal Processing</i> , 25(1):24-33 (1977)
Oppenheim and Schaffer, <i>Discrete-Time Signal Processing</i> , pp. 17-33 (1989)

APPENDIX C

'521 Patent Claim Elements	Ozawa (Ex. 1006) ¹
1. A pitch analysis device for producing a set of pitch codebook parameters, comprising:	“The <u>pitch period of the speech signal is determined</u> from the weighted speech samples.” Ex. 1006, 2:6-7 “In order to determine the pitch period at subframe intervals. the pitch synthesizer 17 is provided with a <u>adaptive codebook</u>.” <i>Id.</i> 4:17-19. <i>See also</i> ¶¶87-88.
a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data,	“The output of the vector selector 22, the <u>pitch index from pitch synthesizer 17</u> and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation: $D_k = \sum_{n=c}^{N-1} X'w(n) - \beta'_k \cdot q(n) - g'1u(n) * h_w(n) - g'2u(n) * h_w(n) ^2 \quad (6)$ where, β'_k is the gain of <u>k-th adaptive code vector</u>.” <i>Id.</i> 5:22-32. <i>See also</i> ¶¶87-88.
wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;	“According to a first aspect of the present invention, <u>an input speech signal is segmented into speech samples</u> at first intervals” <i>Id.</i> 1:66-2:1. <i>See also</i> ¶¶89-90.
a) at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, wherein:	“The weighting function η is therefore varied with the gain code vectors and the comb filter 21 <u>produces for each of its inputs. a plurality of filtered excitation code vectors</u>” <i>Id.</i> 6:46-49 “A modified embodiment of the present invention is shown in FIG. 2 in which the vector selector 22 is connected between the

¹ All underlining in the claim chart denotes emphasis added.

gain calculator 23 and multiplexer 25 to receive its inputs from the outputs of gain calculator 23 and from the outputs of excitation vector candidate selector 18. The vector selector 22 receives its inputs direct from filter 21. Gain calculator 23 makes a search for a gain code vector using a three-dimensional gain codebook 24'0. During mode 1, 2 or 3, vector selector 22 searches for a gain code vector that minimizes Equation (6) with a respect to each of the filtered excitation code vectors, and during mode 0 it searches for one that minimizes Equation (7) with respect to each excitation code vector. Vector selector 22 selects one of the candidate code vectors and one of the gain code vectors that minimize the distortion given Equation (6) during mode 1, 2 or 3, or minimize the distortion given by Equation (7) during mode 0, and delivers the selected candidate excitation code vector and the selected gain code vector as excitation and gain indices to multiplexer 25 as well as to excitation pulse synthesizer 28.” *Id.* 6:21-40. *See also* Fig. 2.



See also ¶¶91-92.

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; and	“The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation: $D_k = \sum_{n=c}^{N-1} [Xw(n) - \beta'_k \cdot q(n) - g'_{1k}c_{1k}(n) \cdot h_w(n) - g'_{2k}c_{2k}(n) \cdot h_w(n)]^2 \quad (6)$ where, β'_k is the gain of <u>k-th adaptive code vector</u>.” Ex. 1006, 5:22-33 “In each operating mode, the gain calculator generates a gain index representing the quantized optimum gain code vector.” <i>Id.</i> 5:40-41. <i>See also</i> ¶¶93-95.
ii) 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; and	“<u>The excitation vector candidates, the pitch index and the mode index are applied to the comb filter 21</u> in which its delay time T is set equal to the pitch period. During mode 1, 2 or 3, each of the excitation code vector candidates is passed through the comb filter Ex. 1006 4:57-61. “Alternatively, a different value of weighting function p may be used for each mode of operation.” <i>Id.</i> 4:67-5:1. <i>See also</i> ¶¶96-97.
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.	“Excitation vector candidate selector 18 is connected to excitation vector codebooks 19 and 20 to search for excitation vector candidates and <u>to select excitation code vectors such that those having smaller amounts of distortion are selected</u> with higher priorities. When the encoder is in mode 1, 2 or 3, it makes a <u>search through the codebooks 19 and 20 for excitation code vectors that minimize the amount of distortion</u> represented by the following Equation:” <i>Id.</i> 4:27-34.

	“Since the excitation code vector candidates are selected in number corresponding to the subframes and filtered through the moving average type comb filter 21, and <u>one of the candidates is selected so that speech distortion is minimized</u>,” <i>Id.</i> 6:14-17. “During mode 1, 2 or 3, vector selector 22 searches for a gain code vector that minimizes Equation (6) <u>with a respect to each of the filtered excitation code vectors</u>, and during mode 0 it searches for one that minimizes Equation (7) <u>with respect to each excitation code vector</u>. Vector selector 22 selects one of the candidate code vectors and one of the gain code vectors that minimize the distortion given Equation (6) during mode 1, 2 or 3, or minimize the distortion given by Equation (7) during mode 0, and delivers the selected candidate excitation code vector and the selected gain code vector as excitation and gain indices to multiplexer 25 as well as to excitation pulse synthesizer 28.” <i>Id.</i> 6:21-40. <i>See also ¶¶98-102.</i>
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.	“A modified embodiment of the present invention is shown in FIG. 2 in which the vector selector 22 is connected between the gain calculator 23 and multiplexer 25 to receive its inputs from the outputs of gain calculator 23 and from the outputs of excitation vector candidate selector 18. The vector selector 22 receives its inputs direct from filter 21. During mode 1, 2 or 3, vector selector 22 searches for a gain code vector that minimizes Equation (6) with a respect to each of the <u>filtered excitation code vectors</u>,

	and <u>during mode 0 it searches for one that minimizes Equation (7) with respect to each excitation code vector.</u>” <i>Id.</i> 6:21-33. <i>See also ¶¶104-106.</i>
5. A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:	<i>See citations to claim 1 above.</i>
a) a convolution unit for convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	“At subframe intervals, excitation pulse synthesizer 28 responds to the spectral parameters α_{ij} and α'_{ij} and LSP index by calculating the following Equation to modify the excitation pulse $v(n)$: $d(n) = \sum_{l=1}^{10} \alpha_l \cdot q(n-l) + \sum_{l=1}^{10} \alpha'_l \cdot g^l \cdot p(n-l) + \sum_{l=1}^{10} \alpha'_l \cdot g^l \cdot d(n-l) \quad (10)$ where $p(n)$ is the output of the weighting filter of the excitation pulse synthesizer.” Ex. 1006, 5:56-67; <i>see also</i> Ex. 1006, 1:57-59. <i>See also ¶¶109-111.</i>
b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	“The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation: $D_k = \sum_{n=c}^{N-1} X'w(n) - \beta'_k \cdot q(n) - g'1u(n) * h_w(n) - g'2u(n) * h_w(n) ^2 \quad (6)$ where, β'_k is <u>the gain of k-th adaptive code vector.</u>” <i>Id.</i> 5:22-32. <i>See also ¶¶112-113.</i>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	“In order to determine the pitch period at subframe intervals. the pitch synthesizer 17 is provided with a known adaptive codebook. During mode 1, 2, or 3. a pitch period is derived from an output sample $X'w(n)$ of subtractor 16 using the impulse

	response $hw(n)$ from impulse response calculator 26.” Ex. 1006, 4:17-22. “ ... β is the gain of the pitch synthesizer 17” Ex. 1006, 4:38-39. “The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art <u>the gain calculator</u> 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation. .”<i>Id.</i> 5:22-30. <i>See also ¶¶114-115.</i>
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 output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 <u>are coupled to a gain search</u> are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector <u>that minimizes distortion</u> represented by the following Equation: $D_k = \sum_{n=c}^{N-1} X'w(n) - \beta'_k \cdot q(n) - g'_{1k}c_{1d}(n)*h_w(n) - g'_{2k}c_{2d}(n)*h_w(n) ^2 \quad (6)$ <i>Id.</i> 5:22-30. <i>See also ¶¶116-120.</i>
6. A pitch analysis device as defined in claim 5, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.	<i>See citations to claim 5.</i> “the gain calculator 23 searches the codebook 24 for a gain code vector <u>that minimizes distortion</u> represented by the following Equation: $D_k = \sum_{n=c}^{N-1} X'w(n) - \beta'_k \cdot q(n) - g'_{1k}c_{1d}(n)*h_w(n) - g'_{2k}c_{2d}(n)*h_w(n) ^2 \quad (6)$ <i>Id.</i> 5:24-30 “β'_k is the gain of k-th 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.” <i>Id.</i> 5:32-34; <i>see also id.</i> 5:34-43. <u>Ex. 1004</u> “Once the adaptive-codebook delay is

	determined, the adaptive-codebook gain g_p is computed as: $g_p = \frac{\sum_{n=0}^{39} x(n) y(n)}{\sum_{n=0}^{39} y(n) y(n)} \quad \text{bounded by } 0 \leq g_p \leq 1.$. . . where $x(n)$ is the target signal and $y(n)$ is the filtered adaptive-codebook vector .” <i>See also ¶¶170-179.</i>
7. A pitch analysis device as defined in claim 5, wherein:	<i>See citations to claim 5 above.</i>
a) each of said filters of the plurality of signal paths is identified by a filter index;	“A mode classifier 27 is connected to the spectral parameter calculator 12 to evaluate the linear predictive coefficients α_{ij}. Specifically, it calculates K-parameters that represent the spectral envelope of the speech samples of every five subframes. A technique described in a paper “Quantization Properties of Transmission Parameters in Linear Predictive Systems”. John Makhoul et. Al, IEEE Transactions ASSP. pages 309 to 321. 1983. is available for this purpose. 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 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 <u>mode index</u> is supplied from the mode classifier to the pitch synthesizer 17 and to the multiplexer 25 for transmission or storage.” <i>Id.</i> 4:1-16. “The excitation vector candidates, the pitch index and the <u>mode index</u> are applied to the

	comb filter 21 in which its delay time T is set equal to the pitch period. During mode 1, 2 or 3, each of the excitation code vector candidates is passed through the comb filter so that, if the order of the filter is 1, the following excitation code vector $C_{jz}(n)$ is produced as a comb filter output: $C_{jz}(n) = C_j(n) + \rho C_j(n-T) \quad (4)$ where $C_j(n)$ is the excitation code vector candidate j, and ρ is the <u>weighting function of the comb filter</u>. Alternatively, <u>a different value of weighting function ρ may be used for each mode of operation.</u>” <i>Id.</i> 4:57-5:2. <i>See also ¶¶122-125.</i>
b) said pitch codevector is identified by a pitch codebook index;	“Pitch synthesizer 17 supplies a <u>pitch index indicating the pitch period to an excitation vector candidate selector 18</u>, a comb filter 21, a vector selector 22, an excitation pulse synthesizer 28 and to the multiplexer 25. When the encoder is in mode 0, the pitch synthesizer produces no pitch index.” <i>Id.</i> 4:22-26 <i>See also ¶¶126-127.</i>
c) said pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain.	“Excitation pulse synthesizer 28 receives the gain index, excitation indices, mode index and pitch index and reads corresponding vectors from codebooks, not shown.” <i>Id.</i> 5:44-55. <i>See also ¶¶128-130.</i>
8. A pitch analysis device as defined in claim 1,	<i>See citations to claim 1 above.</i>
wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error	“The filtered vector candidates $C_{jz}(n)$, the pitch index and the output of subtractor 16 are applied to the vector selector 22. For the first and second excitation vector stages (corresponding respectively to codebooks 19 and 20), the <u>vector selector 22</u> selects those of the filtered candidates which

	minimizes the distortion given by the following Equation: $D_c = \sum_{n=c}^{N-1} X^*w(n) - \beta \cdot q(n) - g_{1c1c}(n) * h_w(n) - g_{2c2c}(n) * h_w(n) ^2 \quad (5)$ 25.” <i>Id.</i> 5:8-21. “The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the <u>gain calculator 23 searches the codebook 24</u> for a gain code vector <u>that minimizes distortion</u> represented by the following Equation: $D_k = \sum_{n=c}^{N-1} X^*w(n) - \beta'_k \cdot q(n) - g'_{1k1k}(n) * h_w(n) - g'_{2k2k}(n) * h_w(n) ^2 \quad (6)$ <i>Id.</i> 5:22-39. <i>See also</i> ¶¶132-137.
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.	<i>See Id.</i> 5:8-21 and 5:22-39, above (“minimizes distortion”) <i>See also</i> ¶¶138-139.
10. An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising:	<i>See citations to claim 1 above.</i> <u>Ex. 1007</u> “Many of the advantages of the well-known CELP coders and decoders are not fully realized when applied to the communication of wide-band speech information (e.g., in the frequency range 50 to 7000 Hz). The present invention, in typical embodiments, seeks to adapt existing CELP techniques to extend to communication of such wide-band speech and other such signals.” Ex. 1007, 1:64-2:2.

	“The prospect of high-quality multi-channel/multi-user speech communication via the emerging ISDN has increased interest in advanced coding algorithms for wideband speech. In contrast to the standard telephony band of 200 to 3400 Hz, wideband speech is assigned the band 50 to 7000 Hz and is sampled at a rate of 16000 Hz for subsequent digital processing. The added low frequencies increase the voice naturalness and enhance the sense of closeness whereas the added high frequencies make the speech sound crisper and more intelligible. The overall quality of wideband speech as defined above is sufficient for sustained commentary-grade voice communication as required, for example, in multi-user audio-video teleconferencing.” Ex. 1007, 1:41-54. “In contrast to the standard telephony band of 200 to 3400 Hz, the wideband speech considered here is characterized by a spectral band of 50 to 7000 Hz. The added low frequencies enhance the naturalness and authenticity of the speech sounds. The added high frequencies make the sound crisper and more intelligible. The signal is sampled at 16 KHz for digital processing by the CELP system. The higher sampling rate and the added low frequencies both make the signal more predictable and the overall prediction gain is typically higher than that of standard telephony speech. The spectral dynamic range is considerably higher than that of telephony speech where the added high-frequency region of 3400 to 6000 Hz is usually near the bottom of this range. Based on the analysis in the previous section, it is clear that, while coding of the
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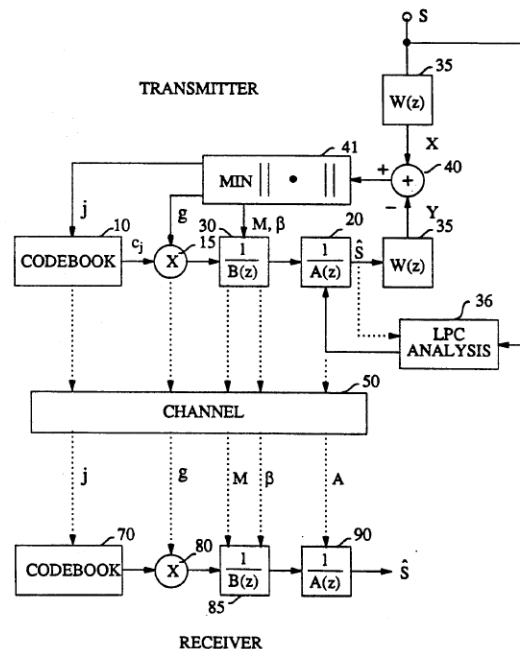
	low-frequency region should be easier, coding of the high-frequency region poses a severe problem. The initial unweighted spectral SNR tends to be highly negative in this region. On the other hand, the auditory system is quite sensitive in this region and the quantization distortions are clearly audible in a form of crackling and hiss. Noise weighting is, therefore, more crucial, in wideband CELP. The balance of low to high frequency coding is more delicate. The major effort in this study was towards finding a good weighting filter that would allow a better control of this balance.” Ex. 1007, 5:40-66. <i>See also ¶¶187-188.</i>
a) a linear prediction synthesis filter calculator responsive to the wideband signal for producing linear prediction synthesis filter coefficients;	“Digital speech signals on each 8-ms subframe are supplied to a perceptual weighting circuit 15 and to a spectral parameter and LSP (SP/LSP) calculation circuit 12 where they are masked by a window of an approximately 24-millisecond length. Computations are performed on the signals extracted through the window to produce a spectral parameter of the speech samples. The number of computations corresponds to the order p (typically, p=10). Known techniques are available for this purpose, such as LPC (linear predictive coding) and the Burg's method. The latter is described in "Maximum Entropy spectrum Analysis", J. P. Burg, Ph.D. dissertation, Department of Geophysics, Stanford University, Stanford, Calif., 1975.” <i>Id.</i> 2:61-3:6. <u>Ex. 1007</u> “Shown are the transmitter portion at the top of the figure, the receiver portion at the bottom and the various parameters (j, g, M,

β and A) that are transmitted via a communication channel 50. CELP is based upon the traditional excitation-filter model where an excitation signal, drawn from an excitation codebook 10, is used as an input to an all-pole filter which is usually a cascade of an LPC-derived filter $1/A(z)$ (20 in FIG. 1) and a so-called pitch filter $1/B(z)$, 30. The LPC polynomial is given by

$$A(z) = \sum_{i=0}^M a_i z^{-i}$$

and is obtained by a standard M^{th} -order LPC analysis of the speech signal.” Ex. 1007, 2:40-55.

FIG. 1



See also ¶¶189-190.

b) a perceptual weighting filter, responsive to the wideband signal and the linear prediction synthesis filter coefficients, for producing a perceptually weighted signal;

“The linear predictive coefficients α_{ij} (where j indicates subframes 1 to 5) are supplied at subframe intervals from the circuit 12 to the perceptual weighting circuit 15 so that the speech samples from the subframing circuit 11 are weighted by the linear predictive coefficients. A series of

	perceptually weighted digital speech samples $X_w(n)$ are generated and supplied to a subtractor 16 . . .” <i>Id.</i> 3:19-25; <i>see</i> Fig. 1, (perceptual weighting 15). <u>Ex. 1007</u> “As seen in FIG. 1, the WMSE matching is accomplished via the use of a noise-weighting filter $W(z)$ 35. The input speech $s(n)$ is first pre-filtered by $W(z)$ and the resulting signal $x(n)$ ($X(z)=S(z) W(z)$) serves as a reference signal in the closed-loop search.” Ex. 1007, 3:6-11. <i>See also</i> ¶¶191-193.
c) an impulse response generator responsive to said linear prediction synthesis filter coefficients for producing a weighted synthesis filter impulse response signal;	“The <u>linear predictive coefficients</u> (where j indicates subframes 1 to 5) are supplied at subframe intervals from the circuit 12 <u>to the perceptual weighting circuit 15</u> so that the speech samples from the subframing circuit 11 <u>are weighted by the linear predictive coefficients.</u>”<i>Id.</i> 3:19-23 “Using the linear predictive coefficients α_{ij} and α'_{ij}, the impulse response calculator 26 <u>calculates the impulse responses $h_w(n)$</u> of the weighting filter of an excitation pulse synthesizer 28 . . . The output of impulse response calculator 26 is supplied to the pitch synthesizer 17 to allow it to determine the pitch period of the speech signal.” <i>Id.</i> 3:56-67. <i>See also</i> ¶¶194-196.
d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:	Ex. 1006, 4:17-23; <i>See</i> Fig. 1-2 (elements 16, 17, 18, 19, 20, 21, 22, 23, 24, 28, and 29) <i>See also</i> ¶¶197-199.
i) said pitch codebook search device responsive to the perceptually weighted signal and	<i>See</i> citations to claim 1 above. “In order to determine the pitch period at

the linear prediction synthesis filter coefficients for producing the pitch codevector and an innovative search target vector; and	subframe intervals, the pitch synthesizer 17 is provided with <u>a known adaptive codebook</u>. During mode 1, 2 or 3, a pitch period is derived from an output sample $X'w(n)$ of subtractor 16 using the impulse response $h_w(n)$ from impulse response calculator 26. Pitch synthesizer 17 <u>supplies a pitch index indicating the pitch period</u>" <i>Id.</i> 4:17-23. "Excitation vector candidate selector 18 is connected to <u>excitation vector codebooks 19 and 20 to search for excitation vector candidates</u> and to select excitation code vectors such that those having smaller amounts of distortion are selected with higher priorities." <i>Id.</i> 4:27-31. <i>See also</i> ¶¶197-199.
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;	<i>See</i> citations to claim 1 above. "The output of the vector selector 22, the pitch index from pitch synthesizer 17 and the output of subtractor 16 are coupled to a gain search are known in the art the gain calculator 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation . . . where, β'_k is the gain of k-th adaptive code vector, and g'_{1k} and g'_{2k} are the gains of <u>k-th excitation code vectors</u> of the first and second excitation vector stages, respectively. During mode 0, the following Equation is used to search for an optimum gain code vector . . . In each operating mode, the gain calculator generates a gain index representing the quantized optimum gain code vector for application to the excitation pulse synthesizer 28 as well as to the multiplexer 25 for transmission or storage." <i>Id.</i> 5:22-43. <i>See also</i> ¶¶200-201.

e) an innovative codebook search device, responsive to a weighted synthesis filter impulse response signal, and the innovative search target vector, for producing innovative codebook parameters;	“Excitation vector candidate selector 18 is connected to excitation vector codebooks 19 and 20 to search for excitation vector candidates and <u>to select excitation code vectors such that those having smaller amounts of distortion are selected</u> with higher priorities. When the encoder is in mode 1, 2 or 3, it makes a <u>search through the codebooks 19 and 20 for excitation code vectors that minimize the amount of distortion</u> represented by <u>the following Equation</u>: “ $D = \sum_{n=0}^{N-1} [Xw(n) - \beta \cdot q(n) - g_1 \cdot c_{1l}(n) * h_w(n) - g_2 \cdot c_{2l}(n) * h_w(n)]^2 \quad (2)$ <i>Id.</i> 4:27-37. <i>See also</i> $h_w(n)$ in equations (2),(3),(6). <i>See also</i> ¶¶202-204.
f) a signal forming device for producing an encoded wideband 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	<i>See</i> citations to claim 10 above. <i>See</i> citations to claim 10 (d) above. <i>See also</i> ¶¶205-214.
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.	<i>See</i> citations to claim 2, above. <i>See also</i> ¶¶217-220.
14. An encoder as defined in claim 10, wherein each pitch prediction error calculating device comprises:	<i>See</i> citations to claim 5, above. <i>See also</i> ¶¶223-229.
a) a convolution unit for convolving the pitch codevector	<i>See</i> citations to claim 5, element 5.a above.

with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	
b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	<i>See citations to claim 5, element 5.b above.</i>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	<i>See citations to claim 5, element 5.c above.</i>
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<i>See citations to claim 5, element 5.d above.</i>
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: $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.	<i>See citations to claim 5, particularly element 5(b) above.</i> <i>See also citations to claim 6.</i> <i>See also Ex. 1006, Figs. 1-2, gain calculation 23 and gain codebook 24.</i> <i>See also ¶¶300-305.</i>
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,	<i>See citations to claim 8 element “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error.”</i> <i>See also ¶¶232-233.</i>

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.	<i>See</i> citations to claim 8, element “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” above. <i>See also</i> ¶¶234-236.
28. A cellular mobile transmitter/receiver unit, comprising:	<u>Ex. 1007</u> “One area of such improvements have come to be called code excited linear predictive (CELP) coders . . . Such techniques have found application, e.g., in voice grade telephone channels, including <u>mobile telephone</u> channels.” Ex. 1007, 1:25-40. <i>See also</i> ¶¶238.
a) 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;	<i>See</i> citations to claim 10. “In each operating mode. the gain calculator generates a gain index representing the quantized optimum gain code vector for application to the excitation pulse synthesizer 28 as well as to the multiplexer 25 <u>for transmission</u> or storage.”<i>Id.</i> 5:40-44 <u>Ex. 1007</u> “Shown are the transmitter portion at the top of the figure, the receiver portion at the bottom and the various parameters (j, g, M, β and A) that are transmitted via a communication channel 50.” Ex. 1007, 2:40-43. “In general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the

	LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. The transmitted information is used at the receiver in well-known fashion to recover the original input information.” Ex. 1007, 3:48-55. <i>See also</i> ¶¶239-242.
and b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding the received encoded wideband signal.	Ex. 1007 “In general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. <u>The transmitted information is used at the receiver in well-known fashion to recover the original input information.</u>” Ex. 1007, 3:48-55; <i>see also id.</i> 5:40-66. <i>See also</i> ¶¶243-246.
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.	<i>See</i> citations to claim 2, above. <i>See also</i> ¶¶249-252.
32. A cellular mobile transmitter/receiver unit as defined in claim 28, wherein each pitch prediction error calculating device comprises:	<i>See</i> citations to claim 5, above. <i>See also</i> ¶¶255-261.
a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and	<i>See</i> citations to claim 5, element 5.a above.

therefore calculating a convolved pitch codevector;	
b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	<i>See citations to claim 5, element 5.b above.</i>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	<i>See citations to claim 5, element 5.c above.</i>
and d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<i>See citations to claim 5, element 5.d above.</i>
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: $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.	<i>See citations to claim 15, above.</i> <i>See also ¶¶307-312.</i>
35. A cellular mobile transmitter/receiver unit as defined in claim 28, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,	<i>See citations to claim 8, element “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” above.</i> <i>See also ¶¶264-265.</i>

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.	<i>See</i> citations to claim 8, element “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,” above. <i>See also</i> ¶¶266-268.
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.	<u>Ex. 1007</u> “One area of such improvements have come to be called code excited linear predictive (CELP) coders . . . Such techniques have found application, e.g., in voice grade <u>telephone channels</u>, including <u>mobile telephone channels</u>.” Ex. 1007, 1:25-40; <i>see also id.</i> 5:40-66. “Shown are <u>the transmitter portion</u> at the top of the figure, the receiver portion at the bottom and the various parameters (j, g, M, β and A) that are transmitted via a communication channel 50.” Ex. 1007, 2:40-43. “In general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g), the pitch lag (p), the pitch tap(s) (β), and the LPC parameters (A). The overall transmission bit rate is determined by the sum of all the bits required for coding these entities. The transmitted information is used at the receiver in well-known fashion to recover the original input information.” Ex. 1007, 3:48-55. <i>See also</i> ¶¶270-276.
38. A network element as defined in claim 37, wherein one of said at least two signal paths	<i>See</i> citations to claim 2, above. <i>See also</i> ¶¶279-282.

comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.	
41. A network element as defined in claim 37, wherein each pitch prediction error calculating device comprises:	<i>See citations to claim 5, above. See also ¶¶285-291.</i>
a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	<i>See citations to claim 5, element 5.a above.</i>
b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	<i>See citations to claim 5, element 5.b above.</i>
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	<i>See citations to claim 5, element 5.c above.</i>
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<i>See citations to claim 5, element 5.d above.</i>
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: $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal	<i>See citations to claim 15, above. See also ¶¶314-319.</i>

paths, and where x is said pitch search target vector and $y^{(i)}$ is said convolved pitch codevector.	
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,	<i>See</i> citations to claim 8, element “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” above. <i>See also</i> ¶¶294-295.
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.	<i>See</i> citations to claim 8, element “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,” above. <i>See also</i> ¶¶296-298.
55. A pitch analysis method for producing a set of pitch codebook parameters, comprising:	<i>See</i> citations to claim 1, above. <i>See also</i> ¶¶142-148.
generating a pitch code vector from a pitch codebook search device based on a digitized input audio data, wherein said digitized input audio data represents an input audio signal that has been sampled and digitized;	<i>See</i> citations to claim 1, element “a pitch codebook search device configured to generate a pitch code vector based on a digitized input audio data” above.
a) in at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data, calculating, for each signal path, a pitch prediction error of said pitch codevector from	<i>See</i> citations to claim 1, element 1.a above.

said pitch codebook search device;	
b) in at least one of said at least two signal paths, filtering the pitch codevector before supplying said pitch codevector for calculation of said pitch prediction error of said at least one signal path;	<i>See citations to claim 1, element 1.b above.</i>
c) comparing the pitch prediction errors calculated in said at least two signal paths, choosing the signal path having the lowest calculated pitch prediction error, and selecting the set of pitch codebook parameters associated to the chosen signal path.	<i>See citations to claim 1, element 1.c above.</i>
56. A pitch analysis method as defined in claim 55, wherein, in one of said at least two signal paths, no filtering of the pitch codevector is performed before supplying said pitch codevector to a pitch prediction error calculating device.	<i>See citations to claim 2, above.</i> <i>See also ¶¶151-153.</i>
59. A pitch analysis method as defined in claim 55, wherein calculating a pitch prediction error in each signal path comprises:	<i>See citations to claim 5, above.</i> <i>See also ¶¶156-161.</i>
a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	<i>See citations to claim 5, element 5.a above.</i>
b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	<i>See citations to claim 5, element 5.b above.</i>
c) multiplying the convolved	<i>See citations to claim 5, element 5.c above.</i>

pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	
d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<i>See citations to claim 5, element 5.d above.</i>
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: $b^{(j)} = x^t y^{(j)} / \ y^{(j)}\ ^2$ where $j=0, 1, 2, \dots, K$, and K corresponds to a number of signal paths, and where x is said pitch search target vector and $y^{(j)}$ is said convolved pitch codevector.	<i>See citations to claim 6, above.</i> <i>See also ¶¶182-185.</i>
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,	<i>See citations to claim 8, element “wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error,” above.</i> <i>See also ¶¶164-165.</i>
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.	<i>See citations to claim 8, element “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,” above.</i> <i>See also ¶¶166-167.</i>