

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 No. 7,260,521

**PETITION FOR *INTER PARTES* REVIEW
OF U.S. PATENT NO. 7,260,521**

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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
Spanias, A., “Speech Coding: A Tutorial Review,” <i>Proceedings of the IEEE</i> , 82(10): 1539-82 (1994)	1010
Federal Coding Standard 1016 (February 14, 1991)	1011
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
Honkanen, T., et al., “Enhanced Full Rate Speech Codec for IS-136 Digital Cellular System,” <i>IEEE</i> , pp. 731-734 (1997)	1013
ETSI - GSM Digital Cellular Telecommunications System; Enhanced Full Rate (EFR) Speech Transcoding (GSM 06.60) (1996)	1014
ITU G.722.2 Series G: Transmission Systems and Media, Digital Systems and Networks (2002)	1015
Gersho, A. and Cuperman, V., “Vector Quantization: A Pattern-Matching Technique for Speech Coding,” <i>IEEE</i> , pp. 15-21 (1983)	1016
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)	1017
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)	1018

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 Vector Quantization in SELP," <i>IEEE</i> , pp. 155-158 (1988)	1020
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
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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
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)	1033
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Pursuant to 35 U.S.C. § 311, ZTE USA, Inc. (“Petitioner”) hereby respectfully requests *inter partes* review of claims 1, 2, 5-8, 10, 11, 14, 15, 17, 28, 29, 32, 33, 35, 37, 38, 41, 42, 44, 55, 56, 59, 60, and 62 of U.S. Patent No. 7,260,521 (“the ’521 Patent”) (Ex. 1001), which issued on August 21, 2007 under 35 U.S.C. §§ 102 and 103. This petition challenges the ’521 patent on grounds previously raised in IPR2015-01875, filed by LG. LG has no control over or interest in the outcome of this proceeding, as IPR2015-01875 was terminated as a result of settlement prior to the Board reaching a decision on institution.

I. MANDATORY NOTICES PURSUANT TO 37 C.F.R. §42.8

A. Real Parties-In-Interest. ZTE Corporation and ZTE (TX) Inc. are real parties-in-interest with Petitioner ZTE USA, Inc.

B. Related Matters—§42.8(b)(2). The ’521 Patent is the subject of a patent infringement lawsuit in the E.D. Tex., No. 2:15-cv-349, *St. Lawrence Commc’n LLC (“SLC”) v. ZTE Corp. et al.* See Ex. 1003. Other U.S. district court actions in which Patent Owner asserted the ’521 patent include E.D. Tex. actions: *SLC v. Samsung Electronics Co. Ltd. et al.*, No. 2:14-cv-293 (closed 4/22/14); *SLC v. LG Electronics, Inc. et al.*, No. 2:14-cv-1055 (closed 1/7/16); *SLC v. Sony Mobile Commc’n (USA) Inc. et al.*, No. 2:15-cv-350 (4/7/15); *SLC v. Motorola Mobility LLC*, No. 2:15-cv-351; *SLC v. HTC Corp. et al.*, No. 2:15-cv-919; *SLC v. Apple Inc. et al.*, No. 2:16-cv-82. Petitioner is concurrently filing an IPR petition against

U.S. Patent No. 7,151,802, which is related to the '521 Patent.

C. Lead and Back-up Counsel.

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D. Service Information—§42.8(b)(4). Petitioner consents to electronic service by email to ZTE-SLC-IPRs@finnegan.com, the above listed e-mail addresses, and hand delivery to the postal mailing address of listed lead counsel.

II. PAYMENT OF FEES (§ 42.103)

The required fees are submitted herewith. The PTO is authorized to charge any additional fees due at any time during this proceeding to Deposit Account No. **06-0916**.

III. STANDING (§ 42.104(a))

Petitioner certifies that the '521 Patent is available for *inter partes* review (IPR) and that it is not barred or estopped from requesting IPR of the '521 Patent.

IV. REQUEST FOR INTER PARTES REVIEW (§ 42.104(b))

Petitioner requests that the Board find claims 1, 2, 5-8, 10, 11, 14, 15, 17, 28, 29, 32, 33, 35, 37, 38, 41, 42, 44, 55, 56, 59, 60, and 62 unpatentable on the grounds explained in Section VIII and its subsections below.

A. Technology Background

The principles of digital speech coding date back to the mid-1970s and nearly all speech coding technology is still based on these underlying principles in which a speech signal is analyzed and split into an excitation source and vocal tract filter component. *See* Ex. 1036, ¶30. As depicted in figure 1 below, the excitation source component represents the periodic vibration of the vocal cords while the vocal tract filter represents the frequency/filtering/shaping of the human vocal tract that gives speech sounds fundamental time and frequency characteristics. *Id.* ¶31.

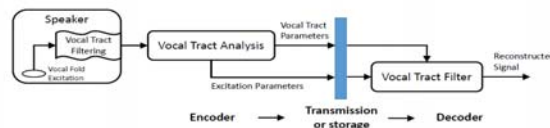


Figure 1 – Source filter model for speech coding and decoding.

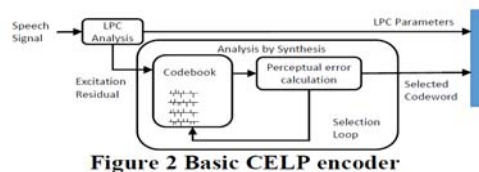
Figure 1 depicts how the source and filter components are separated at the encoder and then compressed separately using as few bits as possible for efficient transmission and storage. The speech signal is re-synthesized at the decoder by first reconstructing the excitation component and then passing it through the vocal tract filter. *Id.* ¶32.

As described in Spanias's 1994 paper, the first step is to divide the signal

into frames or windows and encapsulate a small part of the signal containing approximately one speech component. *Id.* ¶33. Next, typically some initial processing is applied such as a pre-emphasis filter. *Id.* ¶¶33-34. After pre-processing, the representation of the vocal tract filter for each frame is accomplished using Linear Prediction Coding (LPC) analysis and then represented for transmission using LPC coefficients (or other related representations derived from those coefficients). *Id.* ¶35. The LPC coefficients are used to filter each speech signal frame to remove the vocal tract characteristics, giving a residual signal that represents the original excitation component. *Id.* ¶36.

The remaining excitation component is difficult to represent compactly and significantly impacts the quality of the re-synthesized speech thus, the focus has been on finding a good excitation representation. *Id.* ¶37. One of the most common methods first published in 1984 is Code Excited Linear Prediction (CELP). *Id.* ¶38. Figure 2 below depicts Schroeder's 1985 (Ex. 1012) CELP method, which introduces 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, and 2) the use of an analysis-by-synthesis approach in the encoder to select the best codeword. Ex. 1036, ¶39. For vector quantization, a number of different excitation signal vectors are stored in a simple pre-determined list (a codebook) that identifies them by an identifying entry

number (a codeword). *Id.* ¶40. 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. *Id.* Analysis-by-synthesis essentially implements this search by building an extra decoder inside the encoder itself. *Id.* ¶41. 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. *Id.* The best codewords and parameters are then selected to represent the signal, and are transmitted. *Id.*



Variations on the basic CELP approach include the use of perceptual weighting to improve codeword selection quality and the subtraction of a long-term prediction excitation component based on pitch prediction using an adaptive codebook dating back to 1984. *Id.* ¶42. It is important to represent how accurately a listener perceives the reconstructed signal and one of the earliest and easiest ways to do that was by incorporating a simple perceptual weighting filter that emphasizes the frequency components that are most important to speech, prior to calculating error in the analysis-by-synthesis module. *Id.* ¶43.

The excitation component to be represented by the innovation codebook

actually includes several distinct types of excitation components such as the “voiced,” or pitch-related, and “unvoiced” contributions to the signal. *Id.* ¶44. It was known in 1984 that removing the periodic part of the signal corresponding to the pitch prior to using the innovation codebook can substantially reduce error. *Id.* 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 then apply a basic shaping filter. *Id.* ¶¶44-48. 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. *Id.* 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. *Id.* ¶49.

At the decoder, the CELP signal parameters are used to reconstruct each individual signal frame. *Id.* ¶50. 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. *Id.* 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. *Id.* As it was done in 1994, the individual frames are then filtered to undo the initial pre-processing (*e.g.*, pre-emphasis) and connected together for final continuous speech output. *Id.*

B. Alleged Invention Of The '521 Patent

The '521 Patent generally relates to the efficient digital encoding of a wideband speech signal. Ex. 1036, ¶56. The '521 Patent specifically relates to an improved pitch analysis device and method for long-term pitch prediction. *Id.* To accomplish this, the '521 Patent discloses a low-pass shaping filter, selected from one of K pre-determined frequency shaping filters, which is applied to the output of the pitch codebook. *Id.* 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. *Id.* Bits representing the selected filter are transmitted to the decoder along with the other coding information, including the corresponding pitch codevector and gain. *Id.*

V. CLAIM CONSTRUCTION

Petitioner submits that, for purposes of this Petition only,¹ the terms of the '521 Patent should be given their broadest reasonable construction in light of the '521 specification except for those terms discussed below. 37 C.F.R. § 42.100(b).

¹ Petitioner expressly reserves the right to submit constructions for the claims in the related litigation pending in the Eastern District of Texas, under the legal standard applicable in that proceeding, including how a person of ordinary skill in the art would understand the claims.

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 .”); *see also* Ex. 1036, ¶74.

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 ”); 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”); Ex. 1036, ¶75.

The term “means for calculating a pitch gain” is a means-plus-function term subject to interpretation under 35 U.S.C. §112, ¶6.² The broadest reasonable

² Petitioner reserves the right to contend these means-plus-function terms, along

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 $b^{(j)}$ is calculated in a corresponding gain calculator **306**^(j) in association with the frequency shaping filter at index j , using the following relationship: $b^{(j)} = x^t y^{(j)} / \|y^{(j)}\|^2$ ”); 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 calculate the pitch gain using the equation set forth at 12:40-43.” Ex. 1001, 2:31- 37; 12:39-44; *see also* Ex. 1036, ¶76.

The term “means for calculating an energy of the corresponding pitch prediction error” is a means-plus-function term subject to interpretation under 35 U.S.C. §112, ¶6. 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 $e^{(j)}$ for each value of $y^{(j)}$, the value $y^{(j)}$ is multiplied by the gain b by means of a corresponding amplifier **307**^(j) and the value $b^{(j)}y^{(j)}$ is subtracted from the target vector x by means

with certain other claim terms, are indefinite in the district court, but submits for purposes of this Petition that the structure is “a circuit.”

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; *see also* Ex. 1036, ¶77.

The term “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 means-plus-function term subject to interpretation under 35 U.S.C. §112, ¶6. 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 compare 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. Thus, the claimed structure is a circuit. The claimed function of the “means for comparing” is “to use a comparator to compare different values of the energy of the pitch prediction error.” *Id.* 2:31-37; 12:45-47; *see also* Ex. 1036, ¶78.

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

The broadest reasonable construction of “encoded wideband signal” is a wideband signal that has been encoded. Ex. 1001, 9:34-38; Ex. 1036, ¶79.

VI. SUMMARY OF PRIOR ART TO THE '521 PATENT FORMING THE BASES FOR THIS PETITION

A. U.S. Patent 5,797,119 (“Ozawa”)

Ozawa was filed on February 3, 1997 as a file-wrapper continuation of abandoned Ser. No. 08/281,978 filed on July 29, 1994. Ozawa was issued as a patent on August 18, 1998, more than one year before the October 27, 1999 filing date of the PCT Application upon which the '521 Patent is based and is prior art to the '521 Patent at least under pre-AIA 35 U.S.C. § 102(a)-(b). Ex. 1006.

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

G.729 was published at least as of March 1996, more than one year before the October 27, 1999 PCT filing date of the '521 Patent and is prior art to the '521 Patent at least under pre-AIA 35 U.S.C. § 102(b). Ex. 1004, 1 (noting “(03/96)”). G.729 was also copyrighted in 1996, more than one year before the PCT filing date. G.729 discloses the formula calculating the filter gain. Ex. 1004, 19.

C. U.S. Patent 5,235,669 (“Ordentlich”)

Ordentlich issued as a patent on August 10, 1993, more than one year before

the October 27, 1999 PCT filing date of the '521 Patent and is prior art to the '521 Patent at least under pre-AIA 35 U.S.C. § 102(b). Ordentlich discloses coding of wideband speech using a CELP encoder. Ex. 1007, 1:64-2:2.

VII. TERMINOLOGY

Various terms having the same meaning throughout the '521 Patent and the prior art. As such, Petitioner provides the following chart illustrating synonymous terms used throughout the Ozawa reference cited herein to assist in the Board's review of the unpatentability challenges. See Ex. 1036, ¶84.

'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

VIII. GROUNDS FOR UNPATENTABILITY FOR EACH CLAIM

In light of the disclosures detailed below, the '521 Patent is unpatentable for at least the reasons in the following chart and discussed in more detail herein.

No.	Ground	Prior Art	Exhibit Nos.	Claims
1	102(b)	Ozawa	Ex. 1006	1, 2, 5, 7, 8, 55, 56, 59, and 62
2	103(a)	Ozawa and G.729	Ex. 1006 and Ex. 1004	6 and 60
3	103(a)	Ozawa and Ordentlich	Ex. 1006 and Ex. 1007	10, 11, 14, 17, 28, 29, 32, 35, 37, 38, 41, and 44
4	103(a)	Ozawa, G.729 and Ordentlich	Ex. 1006, Ex. 1004, and Ex. 1007	15, 33, and 42

The challenged claims of the '521 Patent have significant overlapping limitations. For example, in many cases the claims are identical but for their preamble, and other claims are identical but for their dependency on different base claims. For these reasons, a single claim chart illustrating where each limitation of the '521 Patent is disclosed in the prior art is provided in Section VII.E. The explanation of why each quotation discloses the corresponding element in the '521 Patent is provided below in Sections VII.A-D. As will be demonstrated in Section VII.A below, Ozawa discloses nearly every limitation of the challenged claims. Grounds 2-4 address discrete limitations not explicitly disclosed in Ozawa.

The claims of Ground 2 all require a gain calculator formula. While this would have been obvious to one of skill in the art in view of Ozawa, for clarity, Petitioner relies on G.729, which explicitly discloses the gain calculator formula. *See* Ex. 1036, ¶168.

The claims in Ground 3 all require a wideband input signal. Again, while it would have been obvious to one of skill in the art to provide a wideband input

signal to the CELP coder of Ozawa, for clarity, Petitioner relies on Ordentlich, which explicitly discloses a wideband input signal. *See* Ex. 1036, ¶188.

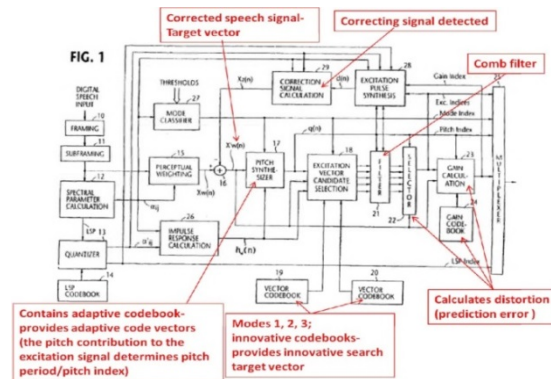
The claims of Ground 4 (Ozawa with G.729 and Ordentlich) require both a wideband signal and a gain calculator formula.

As discussed above in relation to claim construction, some of the challenged claims (e.g., claims 6, 8, 15, 17, 33, 42, 44) contain means-plus-function terms subject to interpretation under 35 U.S.C. §112, ¶6. In relation to such means-plus-function terms, Ozawa discloses functional elements that implement signal processing algorithms expressed by mathematical equations (e.g., equations 1-11). Moreover, 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, the speech coding of Ozawa relies on linear predictive coding (LPC) where “[computations 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 require “computations involved in the gain calculation, excitation pulse syntheses and impulse response calculation on excitation pulses.” Ex. 1006, 6:16-20. A person of ordinary skill in the art would understand that the speech coder of Ozawa incorporates a circuit configured to perform the various computations associated with respective functional elements in accordance with the respective mathematical

equations disclosed by Ozawa.

A. Ground 1: Ozawa Anticipates Claims 1, 2, 5, 7, 8, 55, 56, 59, 62

Ozawa anticipates claims 1, 2, 5, 7, 8, 55, 56, 59, and 62. Ozawa also discloses many of the limitations of claims 11, 14, 17, 29, 32, 35, 41, and 44. For clarity, Ozawa's disclosure of these limitations will be discussed in this section because those claims are materially identical in many respects to the anticipated claims, along with the additional limitations and motivations to combine addressed Patent No. 7,260,521 in Grounds 2 and 3.



Ozawa discloses each and

every limitation of independent claims 1 and 55, which are materially identical. See Ex. 1006, Annotated Fig.1 and corresponding text. The embodiment depicted in Fig. 2 of Ozawa includes the structure depicted in Fig. 1 except for the alternate signal paths at the output of the excitation vector candidate selection 18 to optionally bypass the filter 21. Petitioner relies on the second embodiment depicted in Fig. 2 of Ozawa. Fig. 2 is a modified version of Fig. 1, the first embodiment. Thus, the features other than the modifications depicted in Fig. 2 are common to both embodiments. Petitioner cites to Fig. 1 of Ozawa for those features common to both embodiments and thus no motivation to combine embodiments is necessary

and Ozawa remains a §102 reference.

First, Ozawa discloses a pitch codebook search device to generate a pitch code vector based on a digitized input audio data, which incorporates the adaptive codebook contained in pitch synthesizer 17. Ex. 1006, 4:17-26. The adaptive codebook in the pitch synthesizer 17 provides adaptive code vectors (pitch code vectors). The adaptive codevector from the adaptive codebook is 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 selects a pitch code vector from the adaptive codebook in the pitch synthesizer 17 of Ozawa. Thus, the pitch index provides the pitch code vector of the '521 Patent.

Ozawa also discloses that the digitized input audio data represents an input audio signal that has been sampled and digitized. Ozawa discloses that the input signal is an input audio data because Ozawa discloses digital speech input. Specifically, Ozawa discloses that “an input speech signal is segmented into speech samples at first intervals and a spectral parameter is derived from the speech samples that occur at second intervals longer than the first intervals, the spectral parameter representing the characteristic spectral feature. Each speech sample is weighted with the spectral parameter for producing weighted speech samples.” *Id.* 1:66-2:7. Fig. 1 notes that the input is a “digital speech input” and the signal is sampled and then a spectral parameter is derived from the signal

representing the spectral feature.

Regarding 1.a, Ozawa discloses at least two signal paths associated with respective sets of pitch codebook parameters representative of the digitized input audio data. In Ozawa, 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 two sets of signal paths coming from the pitch synthesizer 17 and the excitation vector candidate selection 18. A first set of signal paths processes the output of the excitation vector candidate selection 18 through the filter 21. Another set of signal paths routes the output of the excitation vector candidate selection 18 directly to the selector 22 bypassing the filter 21.

Regarding 1.a.i, Ozawa also discloses a pitch prediction error calculating device for calculating a pitch prediction error of the pitch codevector from the pitch codebook search device. In Ozawa, the vector selector 22 calculates distortion (prediction error) D of the pitch codevector in equation (5). Ex. 1006, 5:8-21. One of skill in the art would understand that calculating distortion is the same as calculating prediction error. See Ex. 1036, ¶¶93-95. The gain search (gain calculation 23 and gain codebook 24) also calculates a distortion (prediction error) D in equations (6) and (7). Ex. 1006, 5:22-39. In equation 6, $X'w(n)$ 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. One of skill in the art would recognize equation 6 as a calculation of the sum squared prediction error, which includes the long-term prediction component, indicating that Ozawa's distortion calculation is necessarily a pitch prediction error calculation. *Id.*

Regarding 1.a.ii, Ozawa discloses that one signal path comprises a filter 21 for filtering the pitch code vector before supplying it to the pitch prediction error calculating device of the one signal path. Ex. 1006, 4:57-61, 4:67-5:1; *see also* Ex. 1036, ¶¶96-97. In Ozawa, the filter is called comb filter 21 and Ozawa discloses that it receives the excitation vector candidates, the pitch index, and the mode index. *See* Ex. 1006, 4:57-61. Thus, the comb filter 21 filters pitch codevectors specified by the pitch index. Ex. 1036, ¶¶96-97.

Regarding 1.b, Ozawa discloses a selector for comparing the pitch prediction errors calculated in the at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path. Ozawa discloses that 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 1-3 or Equation (7) during mode 0, and delivers those to the multiplexer 25 and the excitation pulse synthesizer 28. *See* Ex. 1006, 6:21-40; Ex. 1036, ¶¶98-102.

In Ozawa, the mode index is determined by the mode classifier 27 which

acts as a switching mechanism and is designed to allow the lowest calculated prediction error. Ex. 1036, ¶¶98-102. The modes, controlled by specified ranges of the overall error, are each designed with pitch prediction filter characteristics (or no filter for mode 0) most well-suited to minimize error within that specific prediction error range, customizing the signal paths for that purpose. *Id.* Then, the excitation codebook selector expressly minimizes the distortion, as described above. *Id.* The net result is to create a system in which the overall prediction error is minimized, and resulting signal quality is maximized. *Id.* Additionally, regardless of operating mode, all required codebook parameters are optimized to minimize prediction error delivered to the multiplexor 25. *See* Ex. 1006, 6:21-40; *see also* Ex. 1036, ¶¶98-102. Claim 55 is the method claim equivalent to apparatus claim 1, and is anticipated by Ozawa for the reasons stated above with respect to claim 1. *See supra* pp. 15-19; *see also* Ex. 1006, 1:17-19, 8:60-11:21, 11:52-12:8, 14:1-21; Ex. 1036, ¶¶142-48.

Ozawa discloses each and every limitation of claims 2, 11, 29, 38, and 56, which are materially identical to each other. Specifically, Ozawa discloses that one of signal paths comprises no filter for filtering the pitch codevector before supplying the 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. *See also* Ex.

1006, 6:21-33; Ex. 1036, ¶¶104-06, 151-53, 217-20, 249-52, 279-82.

Ozawa discloses each and every limitation of claims 5, 14, 32, 41, and 59, which are materially identical to each other. Regarding 5.a, 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. In Ozawa, convolution between the pitch codevector and the weighted synthesis filter impulse response is disclosed in equation (10). Ex. 1036, ¶¶109-11. In equation 10, $q(n)$ is the pitch codevector, $p(n)$ is the weighting filter, and the a_i and a_i' terms are the spectral parameters (Ex. 1006, 5:56-67) that expressly characterize the filter impulse response. Ex. 1006, 1:57-59; Ex. 1036, ¶¶109-11. This convolution equation is disclosed as being implemented within the excitation pulse synthesizer unit 28. *Id.*

Regarding 5.b, Ozawa discloses a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector. In Ozawa, the pitch gain calculator is gain calculator 23. Ozawa discloses that the gain calculator 23 searches a codebook 24 for a gain code vector that includes as a pitch gain the parameter β'_k which is the gain of k -th adaptive code vector. Ex. 1036, ¶¶112-13.

Regarding 5.c, Ozawa discloses an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch

codevector. For example, in equation 6, Ozawa discloses 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. Ozawa discloses a pitch search target vector, denoted the corrected speech signal $X'w(n)$. 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 the convolved pitch codevector $q(n)$, as shown in Fig. 1 (Ex. 1006), which is subtracted from the target vector $X'w(n)$. Ozawa also discloses amplification of the convolved pitch codevector $q(n)$, through the gain parameter β in equations (2), (5), and (6). Ex. 1006, 4:38-39 (“ β is the gain of the pitch synthesizer 17”); Ex. 1036, ¶¶114-115.

Regarding 5.d, 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. Ozawa discloses that the amplified pitch code vector represented by the pitch index $q(n)$ amplified by parameter β is combined with the excitation vector selected by excitation vector candidate selection 18 to calculate the distortion (i.e. pitch prediction error) in equations (2), (5), and (6). Ex. 1036, ¶¶116-20. Each of these equations explicitly discloses the combination of the

convolved pitch code vector and the selected excitation vectors through addition. See 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 *id.* 5:44-6:20; 3:23-31; 4:17-26; see also Ex. 1036, ¶¶116-20.

Ozawa discloses each and every limitation of claim 7. Regarding 7.a, Ozawa discloses each of the filters of the plurality of signal paths is identified by a filter index. 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. Ex. 1036, ¶¶122-125. 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. 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. Ex. 1036, ¶¶122-25. 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. *Id.* Thus, the mode index operates as a filter index that selects a different value of the weighting

function ρ . *Id.*

Regarding 7.b, Ozawa discloses the pitch codevector is identified by a pitch codebook index, which it refers to as the pitch index. Ex. 1006, 4:22-26; Ex. 1036, ¶¶126-27. Finally, regarding 7.c, Ozawa discloses that the “pitch codebook parameters comprise the filter index, the pitch codebook index and the pitch gain” because 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; Ex. 1036, ¶¶128-30.

Ozawa discloses each and every limitation of claims 8, 17, 35, 44, and 62, which are materially identical. Specifically, 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; see Ex. 1036, ¶133. Similarly, the gain calculator 23 calculates the distortion D, which is calculated as a sum of squares of the components of the prediction error vector using equation 6. Ex. 1006, 5:22-39. Also, 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. 1001, 5:32) to the distortion. The distortion or sum-squared value of the pitch prediction error given by equations 5-6 is represents an energy of the pitch prediction error, as understood by a person of ordinary skill in the art. See Ex. 1036, ¶¶136. Moreover, equations 5-6 involve multiplication and

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

Accordingly, these claims are unpatentable under 35 U.S.C. § 102(b) as anticipated by Ozawa, further supported by the claim chart in Section VII.E.

B. Ground 2: Ozawa In View Of G.729 Renders Obvious Claims 6, 15, 42, And 60

Claim 6, depends from claim 5, which depends from claim 1. Ozawa anticipates claims 1 and 5 as discussed above in Ground 1. Ozawa discloses “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 below, G.729 discloses that the pitch gain can be calculated using the claimed equation. 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 the 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.”

1. Disclosure In G.729

G.729 further discloses the formula for calculating the pitch gain required by claim 6. 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. See Ex. 1036, ¶172. G.729 discloses that the adaptive-gain g_p is calculated using the following formula:

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

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. 1036, ¶¶173-74. 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 $b(j)$. *Id.* The equation also uses a multiplier and a subtractor to calculate the mean squared prediction error. The formulas disclosed in G.729 for g_p and $y(n)$ are also identical to the formulas relied on by Patent Owner in its infringement contentions in the related litigation. See Ex. 1034, at 11-12 (claim 6). Thus, G.729 discloses the means for calculating the pitch gain as required by claim 6 and corresponding limitations in similar claims 15, 42, and 60.

2. Motivation To Combine G.729 With Ozawa

Ozawa discloses gain calculation 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. *See* Ex. 1036, ¶176.

G.729 codifies a CELP encoder and decoder structure for coding of speech signals at 8 kbits/s. Although G.729 uses an algebraic CELP (ACELP), ACELP 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. *See* Ex. 1036, ¶177. In fact, G.729 also uses a "filtered adaptive-codebook vector" in its computations. Ex. 1004, 19.

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. *See* Ex. 1036, ¶178.

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, claim 6 of the '521 Patent simply claims a known mathematical formula for calculating the gain of a code vector. Accordingly, these claims are unpatentable under 103(a) as obvious over Ozawa in view of G.729, supported by the chart in Section VII.E.

C. Ground 3: Ozawa In View Of Ordentlich Renders Obvious Claims 10,11,14,17, 28, 29, 32, 35, 37, 38, 41, And 44

Claim 10 depends from claim 1. Ozawa anticipates claim 1 as discussed above in Ground 1. Ozawa discloses the limitations of claims 11, 14, 17, 28, 29,

32, 35, 37, 38 41, and 44 as discussed above. Claims 10, 28, and 37 require a wideband signal. Ozawa does not explicitly disclose that the input speech signal is wideband, however, this is disclosed in Ordentlich as discussed below.

1. Disclosure In Ordentlich

Ordentlich discloses a Low Delay Code-Excited Linear-Predictive Coding of Wideband Speech at 32 Kbits/Sec. That is, Ordentlich discloses encoding the wideband speech using a CELP encoder as required by claims 10, 28, 37, and 38.

First, Ordentlich discloses encoding a wideband input signal because 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 a wideband signal meant that the input signal that is encoded is a wideband signal. See Ex. 1036, ¶188.

Regarding 10.a, Ozawa discloses a linear prediction synthesis for producing linear prediction filter coefficients. Ozawa discloses that “[i]n 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). Ozawa also discloses the perceptual weighting filter of claim 10.b, 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 $X_w(n)$. *Id.* at 3:19-25, *see also* Fig. 1. Next, regarding 10.c, Ozawa 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 $h_w(n)$. *Id.* at 3:56-67. This is supplied to the pitch synthesizer 17 to allow it to determine the pitch period of the speech signal. *Id.*; Ex. 1036, ¶¶189-90.

Regarding limitation 10.d.i, Ozawa discloses a pitch search unit for producing pitch codebook parameters with a pitch codebook search device responsive to the perpetually weighted signal and linear prediction synthesis filter coefficients for producing the pitch codevector and innovative search target vector. 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. *Id.* 4:27-53; Ex. 1036, ¶¶197-99.

Regarding 10.d.ii, the pitch codebook parameters having the lowest calculated pitch prediction error are used. Ozawa 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.**” *Id.* at 5:22-43 (emphasis added); Ex. 1036, ¶¶200-01.

Regarding 10.e, Ozawa discloses that the excitation vector candidate selection 18 is responsive to impulse response signal because it uses the parameters $h(n)$ to calculate the prediction error. *Id.* 4:27-53, Eqs. 2-3, Fig. 1; Ex. 1036, ¶202-04.

Regarding 10.f, 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. *Id.* 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); *see also* Ex. 1036, ¶¶205-14.

Ordentlich also discloses the cellular mobile transmitter/receiver unit of claim 28 and the network element of claim 37 because it discloses that “[s]uch techniques have found application, e.g., in voice grade **telephone channels**, including **mobile telephone** channels.” Ex. 1007, 1:25-40 (emphasis added); Ex. 1036, H238, 270-76. Also regarding claims 28 and 37, Ordentlich discloses a transmission circuit for transmitting the encoded wideband signal because it discloses a transmitter portion (2:40-43) and 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) (P), 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. Regarding claim 28, 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). Thus, claims 28 and 37 are obvious over Ozawa in view of Ordentlich.

2. Motivation To Combine Ordentlich With Ozawa

Ozawa also discloses a CELP codec that adopts a 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 CELP codec also implemented in Ordentlich. See Ex. 1036, 1207.

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 provide a code vector (innovation code vector) that is combined with “segments of past excitation” (pitch code vector) 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; Ex. 1036, ¶208.

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 includes an express suggestion to modify a CELP encoder to encode wideband speech. Ex. 1036, ¶209. 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 (“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) by controlling quantization noise in different frequencies (“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.” *Id.* 5:59-66). Just like Ordentlich, Ozawa discloses using a perceptual weighting filter 15. Ex. 1036, ¶209.

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. *See* Ex. 1036, ¶210. A person of ordinary skill in the art would have been motivated to follow the expressed teachings of Ordentlich to adapt the existing CELP techniques disclosed in Ozawa to extend to communication of wideband speech for at least the reasons explicitly stated in Ex. 1007. *See* Ex. 1036, ¶211. Indeed, Ordentlich discloses positive benefits that a person of ordinary skill in the art would understand as signs of success for using the teachings of Ordentlich. *Id.* 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. A person of ordinary skill in the art would have thus been motivated by the teachings of Ordentlich to apply a digitized wideband speech signal to the coding method of Ozawa. Ex. 1036, ¶211.

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. *Id.* ¶213. The modification of Ozawa with the teachings of Ordentlich simply arranges old elements with each performing the same function it had been known to perform and yields no more than one would expect from such an arrangement. *Id.* 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. *Id.* Accordingly, these claims are unpatentable under § 103(a) as obvious over Ozawa in view of Ordentlich, which is further supported by the claim chart provided in Section VII.E.

D. Ground 4: Ozawa In View Of G.729 And Further In View Of Ordentlich Renders Obvious Claims 15, 33, And 42

Claim 15 depends from claim 14, which depends from claim 10, which depends from claim 1. Claim 33 depends from claim 32, which depends from claim 28, which depends from claim 10, which depends from claim 1. Claim 42 depends from 41, which depends from claim 37, which depends from claim 10, which depends from claim 1.

1. Disclosure In Ordentlich

Ozawa anticipates claim 1 as discussed above in Ground 1. Ordentlich further discloses decoding the wideband speech using a CELP encoder required in claim 10 as discussed in Ground 3. Ozawa in view of Ordentlich disclose the limitations of claims 14, 28, 32, 37, and 41 as discussed in Ground 3.

2. Disclosure In G.729

G.729 discloses the formula for calculating the filter gain as discussed in Ground 2 above. Ozawa in view of G.729 discloses the formula for calculating the gain as required by claims 15, 33, and 42 as discussed above in Ground 2.

3. Motivation To Combine G.729 And Ordentlich With Ozawa

For the reasons in Ground 3, it would have been obvious to combine Ozawa

and Ordentlich. For the reasons in Ground 2, it would have been obvious to combine Ozawa and G.729. The reasons set forth above for Ground 2 also apply to the combination of Ozawa and Ordentlich because this combination does not impact the way pitch gain is calculated. *See* Ex. 1036, ¶304. 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. *See* Ex. 1036, ¶304. 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. Accordingly, claims 15, 33, and 42 are unpatentable under § 103(a) as obvious over Ozawa in view of G.729 and further in view of Ordentlich, further supported by the claim chart in Section VII.E.

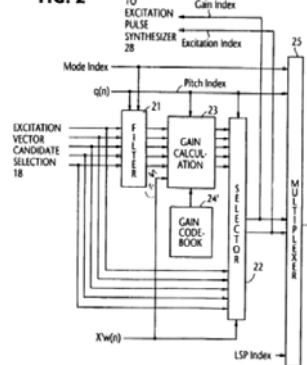
E. Claim Chart Supporting Grounds For Unpatentability

'521 Patent Claim Elements	Ozawa (Ex. 1006) ³
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> Ex. 1036, ¶¶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 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: $D_k = \sum_{n=c}^{N-1} Xw(n) - \beta'_k \cdot q(n) - g'_{1k} c_{1zc}(n) * h_w(n) - g'_{2k} c_{2zc}(n) * h_w(n) ^2 \quad (6)$ where, β'_k is the <u>sain of k-th adaptive code vector</u>.” Ex. 1006, 5:22-32; <i>see also</i> Ex. 1036, ¶¶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, an input speech signal is segmented into speech samples at first intervals” Ex. 1006, 1:66-2:1; <i>see also</i> Ex. 1036, ¶¶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 n is therefore varied with the gain code vectors and the comb filter 21 produces for each of its inputs, a plurality of filtered excitation code vectors” Ex. 1006, 6:46-49 “A modified embodiment of the present invention is shown in FIG. 2 in which the

³ All underlining in the claim chart denotes emphasis added.

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. 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." Ex. 1006, 6:21-40; *see also id. at* Fig. 2.

FIG. 2



See also Ex. 1036, ¶¶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: $D_k = \sum_{n=c}^{N-1} [X'w(n) - \beta'_k \cdot q(n) - g'_{1k}c_{1k}(n) * h_w(n) - g'_{2k}c_{2k}(n) * h_w(n)]^2 \quad (6)$ where, β'_k is the gain of k-th adaptive code vector.” Ex. 1006, 5:22-33 “In each operating mode, the gain calculator generates a gain index representing the quantized optimum gain code vector.” Ex. 1006, 5:40-41; <i>see also</i> Ex. 1036, ¶¶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> Ex. 1036, ¶¶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:” Ex. 1006, 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> at 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> at 6:21-40; see also Ex. 1036, ¶¶98-102
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 filtered excitation code vectors,

	and during mode 0 it searches for one that minimizes Equation (7) with respect to each excitation code vector.” Ex. 1006, 6:21-33; <i>see also</i> Ex. 1036, ¶¶104-106.
5. A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises:	See citations to claim 1 above.
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 $v(n)$: $d(n) = \sum_{i=1}^{10} \alpha_i \cdot q(n-i) + \sum_{i=1}^{10} \alpha'_i \cdot g' \cdot p(n-i) + \sum_{i=1}^{10} \alpha'_i \cdot g' \cdot d(n-i) \quad (10)$ where $p(n)$ is the output of the weighting filter of the excitation pulse synthesizer.” Ex. 1006, 5:56-67; <i>see also id.</i> at 1:57-59; Ex. 1036, ¶¶109-111.
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: $D_k = \sum_{n=c}^{N-1} [X'w(n) - \beta'_k \cdot q(n) - g' \cdot u c_{1H}(n) * h_w(n) - g' \cdot u c_{2L}(n) * h_w(n)]^2 \quad (6)$ where, β'_k is <u>the gain of k-th adaptive code vector.</u>” Ex. 1006, 5:22-32; <i>see also</i> Ex. 1036, ¶¶112-113.
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 $h_w(n)$ from impulse response

	calculator 26.” Ex. 1006,4:17-22. “ ... β is the gain of the pitch synthesizer 17” <i>Id.</i> at 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 the <u>gain calculator</u> 23 searches the codebook 24 for a gain code vector that minimizes distortion represented by the following Equation. . .” <i>Id.</i> at 5:22-30; <i>see also</i> Ex. 1036, ¶¶114-115.
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: $D_k = \sum_{n=c}^{N-1} Xw(n) - \beta_k \cdot q(n) - g'_{1k}c_{1k}(n) * h_w(n) - g'_{2k}c_{2k}(n) * h_w(n) ^2 \quad (6)$ Ex. 1006, 5:22-30; <i>see also</i> Ex. 1036, ¶¶116-120.
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: $D_k = \sum_{n=c}^{N-1} Xw(n) - \beta_k \cdot q(n) - g'_{1k}c_{1k}(n) * h_w(n) - g'_{2k}c_{2k}(n) * h_w(n) ^2 \quad (6)$ Ex. 1006, 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> at 5:32-34; <i>see also id.</i> at 5:34-43. Ex. 1004 “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.” Ex. 1004, 19; <i>see also</i> Ex. 1036, ¶¶170-179.
7. A pitch analysis device as defined in claim 5, wherein:	See citations to claim 5 above.
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.” Ex. 1006, 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> at 4:57-5:2; <i>see also</i> Ex. 1036, ¶¶122-125.
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.” Ex. 1006, 4:22-26; <i>see also</i> Ex. 1036, ¶¶126-127.
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.” Ex. 1006, 5:44-55; <i>see also</i> Ex. 1036, ¶¶128-130.
8. A pitch analysis device as defined in claim 1,	See citations to claim 1 above.
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 selects those of the filtered candidates which minimizes the distortion given by the following Equation:</u>

	$D_c = \sum_{n=c}^{N-1} Xw(n) - \beta \cdot q(n) - g_{1c1g}(n) * h_w(n) - g_{2c2g}(n) * h_w(n) ^2 \quad (5)$ 25.” Ex. 1006, 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 for a gain code vector that minimizes distortion represented by the following Equation:</u> $D_k = \sum_{n=c}^{N-1} Xw(n) - \beta'_k \cdot q(n) - g'_{1k1g}(n) * h_w(n) - g'_{2k2g}(n) * h_w(n) ^2 \quad (6)$ <i>Id.</i> at 5:22-39; <i>see also</i> Ex. 1036, H132-137.
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</i> Ex. 1006, 5:8-21 and 5:22-39, above (“minimizes distortion”) <i>See also</i> Ex. 1036, ¶¶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</i> citations to claim 1 above. <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 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</i> Ex. 1036, H187-188.
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.” Ex. 1006, 2:61-3:6. Ex. 1007 “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

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

The diagram illustrates a transmitter and receiver system for a speech coding method. The transmitter (top) takes an input signal S and processes it through a $W(z)$ block (35) and an adder (+) (40) to produce X . X is then processed by a MIN block (41) to produce B , which is then processed by a $B(z)$ block (15) to produce M . M is then processed by an $A(z)$ block (20) to produce A . A is then processed by an LFC ANALYSIS block (36) to produce S . The receiver (bottom) takes an input signal S and processes it through an $A(z)$ block (85) to produce A , which is then processed by a $B(z)$ block (80) to produce M . M is then processed by a CODEBOOK block (70) to produce J . J is then processed by an adder (+) (80) to produce B , which is then processed by a $W(z)$ block (35) to produce X . X is then processed by an LFC ANALYSIS block (36) to produce S .

“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 . . .” Ex. 1006, 3:19-25; see Fig. 1, (perceptual weighting 15).

“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> Ex. 1036, ¶¶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>” Ex. 1006, 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> at 3:56-67; <i>see also</i> Ex. 1036, ¶¶194-196.
d) a pitch search unit for producing pitch codebook parameters, said pitch search unit comprising:	Ex. 1006, 4:17-23; <i>see id.</i> at Fig. 1-2 (elements 16, 17, 18, 19, 20, 21, 22, 23, 24, 28, and 29); <i>see also</i> Ex. 1036, ¶¶197-199.
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	<i>See citations to claim 1 above.</i> “In order to determine the pitch period at subframe intervals, the pitch synthesizer 17 is provided with a <u>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>” Ex. 1006, 4:17-23. “Excitation vector candidate selector 18 is connected to <u>excitation vector codebooks 19 and 20 to search for excitation vector</u>

	<u>candidates</u> and to select excitation code vectors such that those having smaller amounts of distortion are selected with higher priorities.” <i>Id.</i> at 4:27-31; <i>see also</i> Ex. 1036, ¶¶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;	See 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.” Ex. 1006, 5:22-43; <i>see also</i> Ex. 1036, ¶¶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> : “

	$D = \sum_{n=0}^{N-1} Xw(n) - \beta \cdot q(n) - g_1 \cdot c_1(n) * h_w(n) - g_2 \cdot c_2(n) * h_w(n) ^2 \quad (2)$ Ex. 1006, 4:27-37; <i>see also id.</i>, $h_w(n)$ in equations (2),(3),(6); <i>see also</i> Ex. 1036, ¶¶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	See citations to claim 10 above. See citations to claim 10 (d) above. See also Ex. 1036, ¶¶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.	See citations to claim 2, above. See also Ex. 1036, ¶¶217-220.
14. An encoder as defined in claim 10, wherein each pitch prediction error calculating device comprises:	See citations to claim 5, above. See also Ex. 1036, ¶¶223-229.
a) a convolution unit for convolving the pitch codevector with the weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	See citations to claim 5, element 5.a above.
b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	See citations to claim 5, element 5.b above.
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce	See citations to claim 5, element 5.c above.

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.	See citations to claim 5, element 5.d above.
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.	See citations to claim 5, particularly element 5(b) above. See also citations to claim 6. See also Ex. 1006, Figs. 1-2, gain calculation 23 and gain codebook 24. See also Ex. 1036, ¶¶300-305.
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,	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.” See also Ex. 1036, ¶¶232-233.
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.	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. See also Ex. 1036, ¶¶234-236.
28. A cellular mobile transmitter/receiver unit,	Ex. 1007 “One area of such improvements have come

comprising:	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> Ex. 1036, ¶¶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 citations to claim 10.</i> “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.” Ex. 1006, 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> Ex. 1036, ¶¶239-242.
and b) a receiver including a receiving circuit for receiving a transmitted encoded wideband signal and a decoder for decoding	<u>Ex. 1007</u> “In general, the CELP transmitter codes and transmits the following five entities: the excitation vector (j), the excitation gain (g),

the received encoded wideband signal.	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> Ex. 1036, ¶¶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> Ex. 1036, ¶¶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> Ex. 1036, ¶¶255-261.
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</i> citations to claim 5, element 5.a above.
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</i> citations to claim 5, element 5.b above.
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	<i>See</i> citations to claim 5, element 5.c above.
and d) a combiner circuit for	<i>See</i> citations to claim 5, element 5.d above.

combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	
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.	See citations to claim 15, above. See also Ex. 1036, ¶¶307-312.
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,	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. See also Ex. 1036, ¶¶264-265.
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.	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. See also Ex. 1036, ¶¶266-268.
37. A network element, comprising: a transmitter including an encoder for encoding a wideband signal as recited in	Ex. 1007 “One area of such improvements have come to be called code excited linear predictive (CELP) coders . . . Such techniques have

claim 10 and a transmission circuit for transmitting the encoded wideband signal.	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> Ex. 1036, ¶¶270-276.
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.	<i>See</i> citations to claim 2, above. <i>See also</i> Ex. 1036, ¶¶279-282.
41. A network element as defined in claim 37, wherein each pitch prediction error calculating device comprises:	<i>See</i> citations to claim 5, above. <i>See also</i> Ex. 1036, ¶¶285-291.
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</i> citations to claim 5, element 5.a above.

b) a pitch gain calculator for calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	See citations to claim 5, element 5.b above.
c) an amplifier for multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	See citations to claim 5, element 5.c above.
d) a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	See citations to claim 5, element 5.d above.
42. A network element as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(i)}$ using the relation: $b^{(i)} = x^t y^{(i)} / \ y^{(i)}\ ^2$ 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.	See citations to claim 15, above. See also Ex. 1036, ¶¶314-319.
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,	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. See also Ex. 1036, ¶¶294-295.
and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different	See citations to claim 8, element “wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal

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.	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> Ex. 1036, ¶¶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> Ex. 1036, ¶¶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 said pitch codebook search device;	<i>See</i> citations to claim 1, element 1.a above.
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</i> citations to claim 1, element 1.b above.
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</i> citations to claim 1, element 1.c above.
56. A pitch analysis method as	<i>See</i> citations to claim 2, above.

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 also</i> Ex. 1036, ¶¶151-153.
59. A pitch analysis method as defined in claim 55, wherein calculating a pitch prediction error in each signal path comprises:	<i>See</i> citations to claim 5, above. <i>See also</i> Ex. 1036, ¶¶156-161.
a) convolving the pitch codevector with a weighted synthesis filter impulse response signal and therefore calculating a convolved pitch codevector;	<i>See</i> citations to claim 5, element 5.a above.
b) calculating a pitch gain in response to the convolved pitch codevector and a pitch search target vector;	<i>See</i> citations to claim 5, element 5.b above.
c) multiplying the convolved pitch codevector by the pitch gain to thereby produce an amplified convolved pitch codevector;	<i>See</i> citations to claim 5, element 5.c above.
d) combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error.	<i>See</i> citations to claim 5, element 5.d above.
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	<i>See</i> citations to claim 6, above. <i>See also</i> Ex. 1036, ¶¶182-185.

search target vector and $y^{(i)}$ is said convolved pitch codevector.	
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,	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>See also</i> Ex. 1036, ¶¶164-165.
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.	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>See also</i> Ex. 1036, ¶¶166-167.

Petitioner is likely to prevail with respect to at least one of claims 1, 2, 5-8, 10, 11, 14, 15, 17, 28, 29, 32, 33, 35, 37, 38, 41, 42, 44, 55, 56, 59, 60, and 62 for the reasons given above. Petitioner respectfully requests that the Board grant this petition for *inter partes* review.

Dated: March 10, 2016

Respectfully submitted,

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CERTIFICATE OF SERVICE

The undersigned certifies that the foregoing Petition for *Inter Partes* Review under 35 U.S.C. § 311 and § 18 of the Leahy-Smith America Invents Act, along with the associated Exhibits 1001 through 1036 and Power of Attorney, were served on March 10, 2016 by Express Mail at the following addresses of record for the subject patent:

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The law firm of BIRCH, STEWART, KOLASCH & BIRCH, LLP also has another address listed below; thus, a copy of the aforementioned documents filed on March 10, 2016 were also served at that address via Express Mail.

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A courtesy copy of the same documents were also served at the following address for St. Lawrence's litigation counsel for the subject patent in the infringement lawsuit in E.D. Tex., no. 2:15-cv-349, *Saint Lawrence Communications LLC v. ZTE Corporation, ZTE USA, Inc., and ZTE (TX) Inc.*

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