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UNITED STATES DISTRICT COURT
NORTHERN DISTRICT OF CALIFORNIA
SAN FRANCISCO DIVISION

APPLE COMPUTER, INC., Plaintiff/Counterdefendant, v. BURST.COM, INC., Defendant/Counterclaimant.	§ § § § § § § § §	CASE NO. C06-00019 MHP

DECLARATION OF SHEILA S. HEMAMI IN SUPPORT OF BURST.COM, INC.'S OPPOSITION TO PLAINTIFF APPLE COMPUTER, INC.'S MOTION FOR SUMMARY JUDGMENT ON IVALIDITY BASED ON KRAMER AND KEPLEY PATENTS

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DECLARATION OF SHEILA S. HEMAMI IN SUPPORT OF BURST'S OPPOSITION TO APPLE'S MOTION FOR SUMMARY JUDGMENT OF INVALIDITY BASED ON THE KEPLEY PATENT

I, SHEILA S. HEMAMI, declare that:

1. I am currently a Professor of Electrical and Computer Engineering at Cornell University in Ithaca, New York, where I direct the Visual Communications Laboratory.

2. Appendix A is my curriculum vitae, documenting the details of my professional experience in the areas of digital signal processing and compression. It also lists all of my publications.

3. In summary, I have performed research on the general topics of digital signal processing and compression as a graduate student at Stanford University, as a Member of Technical Staff at Hewlett-Packard Laboratories, and as an Assistant, Associate, and Full Professor of Electrical and Computer Engineering at Cornell University in Ithaca, New York. I have published over 100 refereed journal and conference papers, and I have supervised nine Ph.D. theses, two M.S. theses, and twenty Masters of Engineering projects on topics in digital signal processing and compression. I have taught graduate classes on digital signal processing, digital image processing, and Wiener and Kalman filtering. I also teach in my classes and have a high level of familiarity with image, video, and audio coding standards, including JPEG and JPEG-2000 (standards for still image compression) and MPEG-1/MPEG-2/MPEG-4 (standards for video and wideband audio compression).

4. I am a Senior Member of the Institute for Electrical and Electronics Engineers (IEEE) and a member of the Signal Processing Society (SPS) therein. *Signal processing* broadly deals with the manipulation or processing of an *information representation* (i.e., a signal) to achieve some end goal, such as compression or transmission. Within the IEEE, I have served as Associate Editor for signal representation, coding and compression for the *IEEE Transactions on Signal Processing* journal. In this position, I coordinated reviews and made editorial decisions on papers addressing both theoretical and practical aspects of data compression applied to a multitude of signals, including audio and video. I am currently the elected Chair of the Image & Multidimensional Signal Processing Technical Committee (IMDSP TC) of the IEEE for 2006-8. *Image and multidimensional signal processing* includes not only processing of conventional images and video signals, but also hyperspectral images, data from grids of sensors, and medical

images, to name several examples. The purpose of the IMDSP TC is to promote and guide the advancement of the field of image and multidimensional signal processing, through activities such as coordinating reviews for and sessions within the two major yearly IEEE signal processing conferences (the *International Conference on Acoustics, Speech, and Signal Processing* (ICASSP) and the *International Conference on Image Processing* (ICIP)), nominating community members for technical achievement and service awards, assisting in the selection of SPS Distinguished Lecturers, recruiting associate editors for SPS publications, participating in the development of IEEE standards, and organizing the IEEE's flagship image and multidimensional signal processing conference ICIP. I regularly serve as a reviewer for many IEEE journals and conferences.

5. I have also served on various review panels for the National Science Foundation, to evaluate research proposals for funding; on the program committees of the *Data Compression Conference*, the *Asilomar Conference on Signals, Systems, and Computers*, and *Video Processing and Quality Metrics*; and on the program committees for several conferences organized by the International Society for Optical Engineering (SPIE): *Human Vision and Electronic Imaging* and *Visual Communications and Image Processing*.

6. I am currently providing expert consulting services for Burst.com, Inc. ("Burst") in the lawsuit Apple Computer vs. Burst.com, Inc. I am being compensated at my standard billing rate of $500/hour plus expenses. My compensation is not contingent on the testimony that I intend to offer in this case.

7. My participation in this case to date has included serving as Burst's claim construction expert, including submitting a report in support of Burst's proposed constructions in October 2006, giving a tutorial presentation to the Court in San Francisco on February 1, 2007 prior to the Markman Hearing, and attending the Markman Hearing in San Francisco on February 8, 2007.

8. I have read and studied Burst's four patents — 4,963,995 ("'995"), 5,057,932 ("'932"), 5,164,839 ("'839"), and 5,995,705 ("'705"), which I will collectively refer to as the *Burst patents*. I have also read and studied the Court's claim construction ruling in this case.

9. I have read and reviewed Apple's Motion for Summary Judgement regarding invalidity. I have read and studied Kramer et al. U.S. Patent No. 4,667,088 ("Kramer"), issued May 19, 1987, and Kepley et al. U.S. Patent No. 4,790,003 ("Kepley"), issued December 6, 1988

with a filing date of April 27, 1987. I understand that Apple is arguing that both Kramer and Kepley invalidate the Burst patents.

The Level of Ordinary Skill in the Art

10. Based upon my knowledge of the development of digital communications and networking and my personal involvement in the development of compression technology, a person of ordinary skill in the art at the time that the patent application leading to the ‘995 patent was filed would have had an understanding of: (1) digital communication technologies and their available bandwidths, and (2) audio and/or video compression techniques. In general, a person of ordinary skill in the art would work in the area of digital communication of audio/video source information. A person in this area could be specialized in digital communications having a familiarity with compression technology, or such a person could be specialized in compression technology having a familiarity with digital communications. Such a person of ordinary skill in the art would have had at least a bachelor’s degree in electrical engineering with at least two to three years of experience working on digital communication of audio/video source information. Alternatively, such a person of ordinary skill in the art would have had a master’s degree in electrical engineering with one year of experience working on digital communication of audio/video source information. As another alternative, such a person of ordinary skill in the art would have had a Ph.D. degree in electrical engineering in the area of digital communication of audio/video information.

“Audio/video source information” in Burst’s Claims

11. Kepley teaches a voice mail system for speech, and the Kepley patent operates upon raw material consisting of human speech. That the invention is for speech is made explicitly clear throughout the patent with continuous references to voice in the context of voice mail. Voice mail is a convenience for the conveyance of information, often either in a very terse manner (e.g., “It’s George, call me back at 607-555-1212.”) or in a verbose manner (e.g., “Hi, it’s George, long time no see, maybe we should go golfing some time, I was hoping to catch you but you seem to be out, if you get a chance can you give me a call back at 607-555-1212.”). In neither of these examples nor in the general case is voice mail considered the result of creative talent or creative effort. One of ordinary skill in the art would not consider voice mail to be a “work,” which is part of the Court’s claim construction.

12. In contrast, the Burst patents operate upon raw material consisting of audio and

video in the context of movies or television programs (‘995, 1:6-18, 1:40-62, 2:1-7, 2:18-22, 5:28-32, 7:1-8:2), video only (‘995, 9:48-49), or audio only (‘995, 10:37-41). The operations performed on the material as described in the patents indicate that the information has creative meaning; for example, the video is described as movies or television programs, rather than video from a security system camera. This material — the “audio video source information” of the claims — has been construed to mean “an audio and/or video work that can be received from one or more sources and that has a temporal dimension.”

13. Webster’s dictionary definition of “work” is “something produced by the exercise of creative talent or expenditure of creative effort” which is an appropriate description for the audio/video source information in the Burst patents and an inappropriate description for voice mail messages. [Exhibit A]

14. General audio content such as music and television and movie sound tracks is sometimes called *wideband audio* to emphasize its rich harmonic nature and to delineate it from speech (sometimes called *narrowband audio*). These two types of signals fundamentally differ in their frequency content, which is further described in Paragraph 21. In the remainder of this declaration, use of the word “audio” alone will refer to wideband audio signals such as general audio content described previously. The use of the term “wideband audio” in the late 1980’s was not uniform and was sometimes used to refer to a wider-bandwidth speech signal (this will be discussed further in Paragraph 27 below). [Exhibit I], [Exhibit J]

15. One of ordinary skill would understand that speech and the wideband audio content comprising a “work” as in the Court’s claim construction ruling are different, and that a system designed for speech is not applicable to such works.

16. Speech and wideband audio content would be well understood by one of ordinary skill to be very different types of signals, differing in frequency content and dynamic range, sophistication of the signal, and therefore differing in resulting processing and compression. The subsections below first summarize the Shannon Sampling Theorem (which can be thought of as being included in the “Laws of Physics” and is therefore inviolable) for digitizing an analog signal, which is a necessary precursor to any processing and/or compression operations. Next, the implications of applying this Theorem to speech and to audio are discussed. Finally, the sophistication of a speech signal and of an audio signal are contrasted.

The Shannon Sampling Theorem, or the “Laws of Physics” for Digitization

17. Digitization of an analog signal requires a *sampling rate* (how many samples/ second are required to faithfully represent all of the frequencies present in the signal) and a *resolution* for each sample (how many bits are required to represent each sample). The Shannon Sampling Theorem dictates that the sampling rate (or *sampling frequency*) must be at least twice the maximum frequency present in the signal. Selection of both the sampling rate and the resolution are determined by the characteristics of the signal being processed.

Some Terminology

18. **bits and bytes** — A bit is the smallest unit of digital information and can take on either of two values: 0 or 1. A byte is a unit of information which contains 8 bits. Bits and bytes are related by the equation: number of bytes = number of bits $\times$ 8

19. **Kilobits** — A Kilobit is 1000 bits. It can be abbreviated as Kbit or Kb.

20. **transmission bandwidth, bits/second** — When digital information is transmitted, the speed of transmission (or transmission bandwidth) is measured in bits/second. For example, if 120 bits are received in 60 seconds, the transmission speed is 120 bits/60 seconds = 2 bits/second. Bandwidths can be expressed in Kilobits/second, sometimes abbreviated as Kbps.

Speech and Audio Must be Digitized Using Different Sampling Rates and Resolutions

21. While human hearing spans a frequency range of approximately 5 Hz to 20 KHz, speech is concentrated in the frequency range of 200 Hz to 3.2 KHz or 3.4 KHz, with normal speech having a dynamic range of approximately 30 dB. In contrast, wideband audio content spans the entire frequency range of human hearing — 5 Hz to 20 KHz — with a dynamic range of over 100 dB. “Wideband speech” refers to speech in which the frequency content beyond the “concentrated” region is preserved, from 200 Hz to 7-8 KHz; this bandwidth is slightly greater than twice that of the “concentrated” region. The telephone system in 1988 was designed for speech in the concentrated region and was not designed for wideband speech. [Exhibit I], [Exhibit J]

22. A system designed for speech (such as the telephone system in 1988) could convert the speech to an appropriate digital signal taking 8000 samples/second, and representing each sample with 8 bits. This combination produces bits at a rate of 64 Kilobits/second (the *bit rate* or *data rate*). One of ordinary skill in the art would recognize that the 64 Kilobits/second voice data in Col. 8:30 of Kepley refers to speech sampled in such a manner, and that the system was designed for speech content concentrated below 3.2 KHz. A system designed for wideband audio,

however, typically used at least 40,000 samples/second and 16 bits/sample to convert the audio to an appropriate digital signal. It is possible to use even higher (but not lower) sampling rates in exchange for lower resolution. Apple's brief correctly states "Uncompressed CD-quality audio is 44,100 samples per second and 16 bits per sample, meaning 705,600 bits per second," as such a digitized uncompressed wideband audio signal has a data rate of 705.6 Kilobits/second.

23. Simply sampling a wideband audio signal, such as a top-40 radio song or an aria from an opera, using a system designed for speech processing (at 8000 samples/second with 8 bits/sample) would not work — no useful signal could ever be recovered from those samples because the sampling rate of 8000 samples/second is not at least twice the maximum bandwidth of 20 KHz. A mathematically correct approach would first filter the wideband audio signal to limit its bandwidth to that of speech — but this process would eliminate all frequencies from 3.2 KHz to 20 KHz, nearly 85% of the frequency content of the signal. Clearly, the resulting signal is no longer representative of the original wideband audio signal. Even attempting use of a "wideband speech" system would eliminate 65% of the wideband audio frequencies. Elimination of these frequencies would destroy the acoustic characteristics of the signal which allow a human to recognize and appreciate it as wideband audio (experiments performed in the 1986-8 time frame confirm this and are described later in Paragraph 27). It is akin to calling something the "Mona Lisa" but showing a simplistic cartooned rendition of da Vinci's painting — while the idea of a woman smiling is preserved, the cartoon does not convey the richness and nuances of the painting.

Formation of sounds and therefore Processing/Compression is also different

24. Even after the sampling operation, the processing and compression of digitized speech and digitized wideband audio signals are vastly different. This stems from the fundamental differences in the signals themselves, aside from the basic difference in frequency content discussed above.

25. Speech is formed using the lungs, larynx, and vocal tract. This system is well understood, and many speech processing systems explicitly model this system in designing their processing strategies. Such an approach represents a good engineering design decision — if a system must only handle a specific type of signal, then designing expressly for that signal will provide maximum efficiency and effectiveness.

26. Wideband audio content, however, by virtue of including the entire audible frequency range, includes essentially any sound that a human might hear. It is impossible to create

a model for the infinity of sounds in the world. Even if we limit our discussion to only musical content, consider the vast range of instruments - sound is formed by vibrating strings (e.g., piano or guitar), standing air waves in cavities (e.g., flute), or vibrating surfaces (e.g., percussion instruments). Furthermore, many of them are playing simultaneously. Simultaneous modeling of all possible combinations of instruments is mathematically impossible, and as such taking a model-based approach to wideband audio content is not feasible.

An attempt to code music with a wideband speech coder in 1986-8

27. Standardization efforts for wideband speech signals in the mid-1980's demonstrated that even wideband speech coding approaches were ineffective for music ([Exhibit I]). In 1986 the CCITT adopted G.722, a coding (compression) standard for digital transmission of "wideband audio signals." In this recommendation, "wideband audio" refers to the aforementioned wideband speech, processed using the wider bandwidth of 8 KHz for speech. The standard was aimed toward maximizing speech quality at 64 Kilobits/second. It was evaluated using not only speech but also music, and at 48 Kilobits/second was rated by human listeners as providing quality that was less than good. Note that 48 Kilobits/second is three times that of the claimed compression in Kepley (16 Kilobits/second). One of ordinary skill would recognize that further compression of a 48 Kilobits/second "music" signal to Kepley's 16 Kilobits/second compressed speech signal would destroy the acoustic qualities of the signal.

28. One of ordinary skill would not consider a voice mail message to be the claimed work. Additionally, one of ordinary skill would recognize that a system designed for either narrowband or even wideband speech processing and/or compression would not be suitable for the claimed works.

"Transceiver" in Burst Claims

29. Kepley provides no express teaching that the originating and/or destination voice mail system (e.g., originating voice mail system 110 in Figures 1 and 2) is contained in a common housing. The parties agree that having a common housing is a requirement of the transceiver claim limitations. The voice mail system described in Kepley is shown in Figure 2 consisting of a Voice Storage Processor, a Feature Processor, and a Data Base Processor. One of ordinary skill would understand that while Figure 2 is drawn as three concatenated rectangles, the drawing merely provides ease of understanding the communication paths between the three elements. The Data Base Processor interconnection with the Feature Processor is through an "interface" (290 in

Figure 2) which suggests that separate housings are possible and even likely. Additionally, the Data Base Processor is described as a "system" and as a "machine" (Col. 7:58-60), suggesting a single stand-alone unit. Based on the above, one of ordinary skill in the art would understand that Kepley does not contain clear and convincing evidence of a common housing.

Kepley's Silence Compression Does Not Substantially Lower the Compressed Speech Bit Rate Below 16 Kilobit/second

30. The operative bit rate for the compressed speech contained in voice mail data (which is not a "work") in Kepley is approximately 16 Kilobits/second, because the silence compression discussed in Col. 8:31-35 for "long silences" (Col. 8:32) would not substantially reduce the bit rate.

31. Silence compression was a known approach for reducing the data rate of digital speech in the 1987-88 time frame. Results published in 1988 indicated that 35% of speech data consisted of "silence" of intervals ranging in length from 10 milliseconds to 2.5 seconds ([Exhibit H]). However, 10 milliseconds is clearly not a "long silence" and would not even be considered to be a silence by a human listener. Research performed at AT&T and published in April 1987 reiterated a result published in the Bell Systems Technical Journal in 1965 (the BSTJ was the technical journal of AT&T Bell Laboratories) indicating that 99.56% of "continuous speech" segments have gaps of less than 150 milliseconds in duration ([Exhibit G]). Again, 150 milliseconds would not be considered to be a "long" silence as mentioned in Kepley. As such, one of ordinary skill would conclude that the 35% figure includes a large number of very very short silence intervals, with very few long silences. It would therefore be expected that the operative data rate for the compressed speech would be very close to the 16 Kilobits/second mentioned in Kepley.

32. Common sense when considering not only voice mail messages but ordinary speech intuitively confirms the AT&T result. Long silences only occur when a speaker deliberately pauses for effect or to gather his or her thoughts. From our everyday experiences, voice mail messages are left for a reason, and the speaker seldom has need to gather his or her thoughts because the point of the message is already known to the speaker.

33. In conclusion, one of ordinary skill would understand that the compressed speech bit rate disclosed in the Kepley patent is close enough to 16 Kilobits/second that it can be accurately approximated by 16 Kilobits/second.

Faster-than-Real-Time Transmission in Burst Claims

34. One of ordinary skill would understand that Kepley does not disclose faster-than-real-time transmission of speech, which is not a "work." In the discussions below, I use the set of protocols described in Kepley in conjunction with Figures 4-6 of Kepley to explain this understanding.

35. The high-level transfer of a voice mail message as a computer-to-computer data file transfer is described at Cols. 5:66-6:7. In this paragraph, the transfer is generically described as extending from the originating voice mail system 110 (e.g., the composite system 110 in Figure 1) over a communication line 104 through the public switched telephone network. The network is described as telephone switching system 100, central switching office 130, telephone switching system 140 and to another voice mail service system 150 over a single line 154 (154 and 104 are similar; 100 and 140 are similar; 110 and 150 are similar).

36. The high-level description in the previous Paragraph is further refined in the specification subsection titled "Voice Mail Message Forwarding" which in conjunction with Figure 6 discloses digital transmission of the voice mail message. This subsection describes the transmission of stored voice mail messages in digital format at Col. 12:46-13:37. The digital data is transmitted from "data port circuit 115 of voice mail service system 110" as illustrated in Figure 2, where data port circuit 115 is connected to line 104. The data call connection is then more generally described as advantageously occurring over high speed digital facilities (Col. 13:31-33).

37. Figure 6 is described as illustrating a "message format" (Col. 4:18) and is described under the subheading in the specification "Voice Mail Message Transmission Protocols" (Col. 14:16 - Col. 15:32). A *protocol* is a set of conventions governing the treatment and especially the formatting of data in an electronic communications system ([Exhibit B]). Figure 6 and the associated description illustrate and describe the protocols used in the transmission of the digital voice mail message over a digital communication link, and as such Figure 6 clearly indicates that a digital connection existed between voice mail systems. In particular, DCP stood for "Digital Communication Protocol" and was a proprietary AT&T physical connection which provided the mechanical and electrical connection for digital communication used in their telephony equipment (i.e., the cable description and the electrical specifications for the cable). DCP also included a signaling format which ensured reliable reception of the digitally transmitted information, which is illustrated in Figure 6 as the leftmost rectangle. DCP was for digital

signaling, not analog signaling using a modem. Data port circuit 115, through which the voice mail message is transmitted, is explicitly called a DCP port at Col. 16:7. [Exhibit K]

38. With the digital communication disclosed in Kepley, a person of ordinary skill would understand that faster-than-real-time transmission of compressed voice mail messages as described in the specification was impossible. The combination of the protocols described, the associated overhead with the protocols (e.g., the additional bits required for formatting the information for digital transmission), and the additional overhead shown in Figure 5 specific to the voice mail message preclude faster-than-real-time transmission of a voice mail message coded at 16 Kilobits/second.

39. In the analysis below, I use a bit rate of 16 Kilobits/second for the raw data to be transmitted shown in the right rectangle in Figure 4, labeled "Voice Mail Message." My reasons for this are described in Paragraph 30 above. While the "Address Header" of the left rectangle in Figure 4, which contains information consisting of two phone numbers and a name in text form, would increase the data right above 16 Kilobits/second, the increase would be very small because that information is appended at the beginning of the message, once. This is in contrast with the information appended as shown in Figures 5 and 6, which is added to every 128-byte segment of the entire voice mail message and described below.

Overhead Associated with Transmitting a Voice Mail Message According to Kepley

40. Each protocol described in Kepley — DCP mode 3, LAPD, X.25, UUCP, and MTA — has a "protocol field" associated with it illustrated in Figure 6 as described at 15:7-32. Each of these fields is described as being added to the information of Figure 5, which contains 128 bytes of the voice mail message of Figure 4, plus 4 additional bytes of information. It is important to understand that each of these "protocol fields" consists of a non-zero number of bytes of information which is added to each 128-byte segment of the voice mail message. Because this is additional information which is not part of the actual voice mail message, it is commonly referred to as "overhead" information. While it carries no voice mail information, it is required in order to accurately convey the digital message from one voice mail system to another.

41. LAPD stood for "Link Access Procedure on the D-channel" and referred to the protocol defined in the ISDN[1] standard for the D-channel [Exhibit F]. In 1988, AT&T's D-channel was a 16 Kilobits/second link [Exhibit K]. The LAPD protocol was responsible for providing robust transmission for the subsequent layers. As such, its protocol field was 7-8 bytes long and

included an error-detection capability. This capability is important because Kepley teaches that the digitized voice mail data is transmitted like a computer file. As such, it's important to maintain data integrity, because even single bit errors can be disastrous. While LAPD could detect errors, it could not correct them, and retransmission would be necessary.

42. X.25 is a layer 3 protocol which provides for "data transfer packets" with a header of 3 bytes [Exhibit C].

43. UUCP is a UNIX-based file transfer protocol developed at Bell Laboratories in the 1970's and included a header of 6-7 bytes [Exhibit E]. While there was a version of UUCP designed for use with X.25 that did not have the aforementioned header, that version was designed for text messages and was extremely inefficient for binary data such as Kepley's compressed voice mail messages and one of ordinary skill would not select that version for Kepley's system.

44. MTA refers to Message Transport Architecture and describes apparently proprietary AT&T protocols for their voice mail systems. The MTA header would have a minimum of 2 bytes, at least one for each of the application and presentation layer protocols as described in the specification.

45. Kepley notes that "All of the headers illustrated in Figure 6 are added to the composite data message of Figure 5 for each 128 byte segment of the voice mail message that is transmitted between the two voice mail service systems." (Col 15:28:32). A summary of the overhead information follows:

- LAPD includes 7-8 bytes,
- X.25 includes 3 bytes,
- UUCP includes 6-7 bytes,
- MTA would have a minimum of 2 bytes,
- and Figure 5 discloses 4 bytes appended to the compressed voice mail message.

As such, the number of additional bytes is at minimum 22-24, yielding an overhead of 17.2% to 18.8% $(100 \times 22/128 = 17.2\%$ and $100 \times 24/128 = 18.8\%)$ and an effective bandwidth *for*

1 In 1988, planning of the Integrated Services Digital Network (ISDN) was well underway and the network was highly anticipated in the communications community. ISDN was to be a telephone network in which the internal network and the local loops (i.e., the lines going to residences) were entirely digital, thereby allowing voice traffic and data to be carried in an integrated fashion. LAPD was a protocol designed for the signalling (or "D") channel in ISDN systems.

real-time transmission for the information to be transmitted of 16 Kilobits/second $\times$ (1.172 to 1.188) = 18.8 Kilobits/second to 19 Kilobits/second. Of these numbers, 18.8 Kilobits/second is strictly a lower bound because the overhead associated with MTA may well have been greater than the minimum of 2 bytes assumed in the above calculation. In other words, the required transmission bandwidth is at least and may exceed 18.8 Kilobits/second *in order to achieve real-time transmission*.

46. Any bit errors resulting in retransmission of information would further increase the required transmission bandwidth beyond those values calculated in Paragraph 45.

47. Faster-than-real-time transmission would require transmission bandwidths *exceeding* those for real-time transmission.

48. The combination of the X.25 protocol running under the LAPD protocol, as disclosed in Figure 6 and the accompanying text, provided a default bandwidth of 9.6 Kilobits/second for the voice mail transmission ([Exhibit D]). Kepley does not disclose any deviation from this default protocol combination. This 9.6 Kilobits/second of user data would have to include the UUCP and MTA headers of Figure 6, the headers in Figure 5, and of course the 128 bytes of actual voice mail data. These headers were 12-13 bytes in length, yielding an overhead of 9.4% to 10.2% ($100 \times 12/128 = 9.4\%$ and $100 \times 13/128 = 10.2\%$) and *an effective bandwidth for the information to be transmitted over the 9.6 Kilobits/second link* of 17.5 Kilobits/second to 17.6 Kilobits/second. With a 9.6 Kilobits/second bandwidth for transmission, clearly there was no faster than real time transmission possible and in fact the transmission would be slower than real time.

49. While Kepley states "The use of digital high speed transmission facilities of speed greater than 9.6 Kilobits/second enables the exchange of digitally encoded and compressed voice mail messages faster than real time speech" at Col 13:33-37, clearly selection of *any* speed greater than 9.6 would not provide the claimed delivery. Rather, even a 16 Kilobits/second speed would be insufficient given the required overhead. Furthermore, the selected protocol combination of LAPD and X.25 provided a 9.6 Kilobit/second link.

50. Though Kepley provides no mechanism by which digitized wideband audio signals can be input to the voice mail system, even if such a mechanism existed no faster than real time transmission of the compressed signal would be possible. Applying Kepley's 4:1 compression ratio to a digitized uncompressed wideband audio signal at 705,600 bits/second (Paragraph 22)

results in a data rate of 705,600/4 = 176,400 bits/second. Such a compressed signal would be transmitted slower than real time over any digital communication link of bandwidth less than 176,400 bits/second.

The 19.2 Kilobits/second Link in Kepley Does Not Carry the Voice Mail Messages

51. While "messages" are described in Cols. 9:34-12:45, these are control messages internal to the public switched telephone network and not user-created messages such as voice mail or text messages described elsewhere in the specification. These control messages do not travel over the same lines over which either analog voice calls or data calls, such as the digital voice mail message, travel. These control messages, for example, instruct the voice mail service system to "pick up the phone" when a user decides to create a voice mail message. Col. 8:45-55 describes exactly this situation — the voice mail service system only "answers" when it receives instructions to do so over data link 105. Kepley teaches that these internal system control messages may be communicated between elements of the network by a 19.2 Kilobits/second link. This link does not transmit the digital voice mail messages.

"Digital Input" in Burst Claims

52. Kepley's system is specifically designed to process and compress input speech signals which are analog rather than digital. The voice processor 220 in Figure 2 is described as "where the voice message is converted to digitally encoded voice signals." One of ordinary skill would understand this operation to be analog-to-digital conversion and possible compression. No provisions are given for reception of already-digitized voice signals by the Voice Storage Processor of Figure 2.

53. As such, one of ordinary skill would understand that Kepley does not meet Claim 9 of the Burst '995 and '839 patents, in which the operations of claim 1 — reception, compression, storage, and transmission — are performed on an input signal which is received in digital form.

"Computer Generation" in Burst Claims

54. Furthermore, Kepley's system is for speech, to be left in voice mail messages. As such, the input information is not computer-generated but is rather human-generated, and Claim 15 of the Burst '995 and '839 patents is not met.

Faster than Real Time Retransmission in Burst Claims

55. Kepley teaches a system in which a human sending a voice mail message designates a time for message delivery to the recipient (Col. 12:48-52). No mechanism for

retransmission of a received voice mail message (i.e., forwarding of a voice mail message) is taught, and how such a task would be performed is not clear based on the system described in Kepley. There is simply no discussion about implementation or execution of retransmission in Kepley. For example, header construction as taught in Figure 4 is unclear for a forwarded message — is the old header used with the information of the original sender and receiver; is a new header created with the information of the original receiver (now the sender) and the new receiver, thereby eliminating all information from the original ending; or is some sort of composite header created? Is the original voice mail message including its header information from Figure 4 simply transmitted immediately following a new message which may or may not consist only of a header from the original receiver (now the sender)? One of ordinary skill would ask these and other questions when considering how a forwarding operation might be implemented, and the specification does not provide any guidance whatsoever on this topic. Because the specification explains in great detail how the message is formatted and transmitted, the omission of any retransmission discussion suggests that such retransmission was not envisioned by the inventor.

References Cited

[Exhibit A] *Webster's New Collegiate Dictionary* (1981).

[Exhibit B] Webster's on-line dictionary (`http://www.m-w.com`).

[Exhibit C] R. J. Deasington, *X.25 Explained*, Second Edition, Ellis Horwood Limited, England, 1986.

[Exhibit D] Michel Smouts, *Packet Switching Evolution from Narrowband to Broadband ISDN*, Artech House, 1991.

[Exhibit E] UUCP protocol: `http://www.faqs.org/faqs/uucp-internals/section-7.html`

[Exhibit F] ITU-T Recommendation Q.921, ISDN User-Network Interface-Data Link Layer Specification, 1994. This specification was adopted in 1988 and amended in 1993. The 1988 version is available at `http://www.nmedia.net/docs/ccitt/1988/6_10/6_10_01.txt` where sections 2.1-2.7 which describe the protocol overhead are identical to those in the later document.

[Exhibit G] J. F. Lynch, J. G. Josenhans, R. E. Crochiere, "Speech/Silence Segmentation for real-time coding via rule based adaptive endpoint detection," *Proceedings IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1987.

[Exhibit H] C. K. Gan, R. W. Donaldson, "Adaptive silence deletion for speech storage and voice mail applications," *IEEE Transactions on Acoustics, Speech, and Signal Processing*, Vol. 36, No. 6, June 1988.

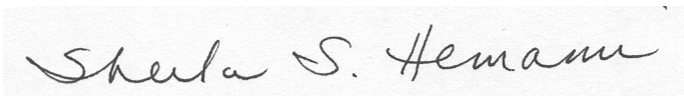
[Exhibit I] Paul Mermelstein, "G.722, a New CCITT Coding Standard for Digital Transmission of Wideband Audio Signals," *IEEE Communications Magazine*, January 1988.

[Exhibit J] Allen Gersho, "Advances in Speech and Audio Compression," *Proceedings of the IEEE*, June 1994.

[Exhibit K] Lee Goeller, "Comparing PBXs For Voice-Data Integration," Business Communications Review, 1988.

I declare under penalty of perjury under the laws of the United States of America that the foregoing is true and correct, and that, based on my knowledge, qualifications, and experience, I am prepared to testify on the subjects addressed herein.

EXECUTED on June 7, 2007 at Ithaca, New York.



Sheila S. Hemami, Ph.D.

Appendix A

Curriculum Vitae of Sheila S. Hemami

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Education

Stanford University — Stanford, CA
Ph.D. in Electrical Engineering, December 1994.

Stanford University — Stanford, CA
M.S. in Electrical Engineering, April 1992.

University of Michigan — Ann Arbor, MI
B.S. in Electrical Engineering, Summa Cum Laude, May 1990.

Professional Experience

Cornell University — Ithaca, NY
Professor, School of Electrical & Computer Engineering, November 2006-present
Associate Professor, School of Electrical & Computer Engineering, November 2000-October 2006.
Assistant Professor, School of Electrical Engineering, January 1995 - October 2000.

Hewlett Packard Laboratories — Palo Alto, CA
Member of Technical Staff, February-December 1994, Interactive Video Initiative: Multimedia Systems & Community Networking Group.

Visiting Positions: Visiting Fellow, Princeton University Dept. of Electrical Engineering, Fall 2001; Texas Instruments Associate Professor of Electrical Engineering, Rice University, Spring 2002.

External Research Funding

Sponsored Research

National Science Foundation, "ACCEL: Advancing Cornell's Commitment to Excellence and Leadership (Cornell University ADVANCE Proposal)," $3,300,000, November 2006-October 2011 (1 of 5 Co-PIs).

National Science Foundation, "A Signal Processing. Approach to Modeling Visual Masking," $255,873, July 2005-June 2008 (sole PI)

National Science Foundation, "A Framework for Encoding American Sign Language and Other Structured Video," joint with University of Washington, $269,584 for Cornell, July 2005-June 2008 (1 of 3 Co-PIs)

National Science Foundation, "Workshop Series on Starting Successful Faculty Careers," $20,000, September 2004-December 2005 (1 of 2 Co-PIs)

National Science Foundation, "Nomination of Professor Robert M. Gray for a Presidential Award for Excellence in Science, Mathematics, and Engineering Mentoring," $10,000 (to Stanford, matched by Stanford to fund a mentoring workshop) (1 of 4 Co-PIs)

Department of the Navy, "Channel and QOS Adaptive Wireless Multimedia Ad-hoc Networks," $4,500,000, 1999-2004 (1 of 7 Co-PIs)

Department of Energy, "Visually Optimized Scalable Image Compression," May 1999-April 2002, $309,000 (sole PI)

National Science Foundation, "Visually Optimized Supra-threshold Image Compression," May 1999-April 2002, $205,000 (sole PI)

National Science Foundation Career Award, "Robust Visual Communications for Packet Networks," May 1997 - April 2001, $209,000 (sole PI)

Department of Energy, "Visual Communications for Heterogeneous Networks," September 1995 - August 1998, $375,000 (sole PI)

National Science Foundation, "A Next Generation Computing and Communication Substrate," July 1997 - June 2002, $1,000,000 (1 of 16 Co-PIs)

Corporate Sponsors

Lockheed-Martin, Tektronix Corporation, Eastman Kodak Company, AT&T, GTE, Intel, Center for Electronic Imaging & Science (New York State)

Memberships

IEEE (Senior Member), Tau Beta Pi, Eta Kappa Nu

Graduate fields of electrical engineering, computer science, and applied math at Cornell

Professional Service Activities

Chair, Image & Multidimensional Signal Processing Technical Committee (IMDSP TC) of the IEEE, 2006-7; general member, October 2001-present (includes program committee duties for ICASSP & ICIP); Awards Subcommittee Chair, 2005

Technical Co-Chair, IMDSP Workshop on Image and Video Quality, June 2007

Program Committee Member: Human Vision and Electronic Imaging, 2007; Video Processing and Quality Metrics, 2006-present; Visual Communications and Image Processing, 2006; Asilomar Conference on Signals, Systems, and Computers, 2005; Data Compression Conference, 2003-present

Organizing/Program Committee, PAESMEM/Stanford School of Engineering Workshop on Mentoring in Engineering, June 2004

Associate Editor, *IEEE Transactions on Signal Processing*, August 2000-April 2006

IEEE International Conference on Image Processing — 2002 Publicity Chair

Reviewer: *National Science Foundation*, *IEEE Transactions on Signal Processing*, *IEEE Transactions on Image Processing*, *IEEE Transactions on Circuits & Systems for Video Technology*, *IEEE Transactions on Communications*, *IEEE Proceedings*, *Journal of Visual*