IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re U.S. Patent No. 9,179,005

Currently in Litigation Styled:
VoIP-Pal.com, Inc. v. Apple Inc.
Case No: 2:16-cv-00260-RFB-VCF

Issued: November 3, 2015

Application Filed: August 13, 2013

Applicant: Clay Perreault, et al.

Title: Producing Routing Messages for Voice Over IP Communications

PETITION FOR *INTER PARTES* REVIEW PURSUANT TO 37 C.F.R. §42.100 ET SEQ.

Mail Stop *Inter Partes* Review
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

DECLARATION OF HENRY H. HOUH, PhD

I. Introduction

I, Henry H. Houh, Ph.D., declare:

1. I am making this declaration at the request of Apple Inc. in the matter of the *Inter Partes* Review of U.S. Patent No. 9,179,005 ("the '005 Patent") to Perreault, *et al*.

2. I am being compensated for my work in this matter. My compensation in no way depends upon the outcome of this proceeding.

3. In the preparation of this declaration, I have studied:

Voip-Pal Ex. 2016
IPR2017-01399

(1) The ’005 Patent, Exhibit 1001;

(2) The prosecution history of the ’005 Patent, Exhibit 1002;

(3) U.S. Patent No. 7,486,684 to Chu et al. (“Chu ’684”), Exhibit 1003;

(4) U.S. Patent No. 8,036,366 to Chu (“Chu ’366”), Exhibit 1004;

(5) U.S. Patent Publication No. 2007/0064919 to Chen et al. (“Chen”), Exhibit 1005;

4. In forming the opinions expressed below, I have considered:

(1) The documents listed above,

(2) The relevant legal standards, including the standard for obviousness provided in *KSR International Co. v. Teleflex, Inc.*, 550 U.S. 398 (2007), and

(3) My knowledge and experience based upon my work in this area, as described below.

II. Qualifications and Professional Experience

5. My complete qualifications and professional experience are described in my curriculum vitae, a copy of which can be found attached hereto as Appendix A. The following is a brief summary of my relevant qualifications and professional experience.

6. I received a Ph.D. in Electrical Engineering and Computer Science from the Massachusetts Institute of Technology in 1998. I also received a Master of Science degree in Electrical Engineering and Computer Science in 1991, a Bachelor of Science Degree in Electrical Engineering and Computer Science in 1989, and a Bachelor of Science Degree in Physics in 1990.

7. As further indicated in my C.V., I have worked in the electrical engineering and computer science fields, including in streaming audio and video, on several occasions. As part of my doctoral research at MIT from 1991-1998, I worked as a research assistant in the Telemedia Network Systems (TNS) group at the Laboratory for Computer Science. The TNS group built a high speed gigabit network and applications which ran over the network, such as remote video capture, processing, segmentation and search on computer terminals. In addition to helping design the core network components, designing and building the high speed links, and designing and writing the device drivers for the interface cards, I also set up the group's web server, which at the time was one of the first several hundred web servers in existence.

8. I authored or co-authored twelve papers and conference presentations on our group's research. I also co-edited the final report of the gigabit networking research effort with the Professor (David Tennenhouse) and Senior Research Scientist of the group (David Clark), who is generally considered to be one of the fathers of the Internet Protocol.

9. I started building web servers in 1993, having set up the web server for the MIT Telemedia, Networks, and Systems Group, to which I belonged. It was one of the first several hundred web servers in existence, and went on to provide what was likely one of the first live Internet video sessions initiated from a web site. I co-authored papers on our web server video system and on database-backed web sites for which I attended the first World Wide Web conference to present.

10. From 1997 to 1999, I was a Senior Scientist and Engineer at NBX Corporation, a start-up that made business telephone systems that streamed packetized audio over data networks instead of using traditional phone lines. NBX was later acquired by 3Com Corporation, and the phone system is still available and being used at tens of thousands of businesses or more. As part of my work at NBX, I designed the core audio reconstruction algorithms for the telephones, as well as the packet transmission algorithms. I also designed and validated the core packet transport protocol used by the phone system. The protocol is used millions of times daily currently. Two of the company founders and I received US Patent No. 6,967,963 titled "Telecommunication method for ensuring on-time delivery of packets containing time sensitive data," for some of the work I did there.

11. Starting in 2001, I was architect for the next generation of web testing product by Empirix known as e-Test Suite. e-Test Suite is now owned by Oracle Corporation. e-Test provided functional and load testing for web sites. e-Test emulated a user's interaction with a web site and provided web developers with a method of creating various scripts and providing both functional testing (*e.g.*, did the web site provide the correct response) and load testing (*e.g.*, could the web site handle 5000 users on its web site simultaneously). Among Empirix's customers was H&R Block, who used e-Test Suite to test the tax filing functionality of their web site as whether the web site could handle a large expected load prior to the filing deadline.

12. At Empirix, I also conceived, secured internal funding for, and managed the engineering for a new data platform test product known as the PacketSphere. The first capability the PacketSphere provided was to emulate a network so that lab testing could be done under conditions that mimicked the Internet, including configurable latency and packet loss. Later, PacketSphere provided the capability to generate large numbers of Voice-over-IP streams as well as measure the quality of the connection of VoIP streams. As part of my work, I continued to study the development of the Voice-over-IP market and worked with a number of Empirix customers to understand their market and product testing needs. Sonus Networks, a leading manufacturer of Voice-over-IP equipment, was a long-time customer of Empirix and one of the first customers of the PacketSphere product.

13. Around 2006, at BBN, I helped create a search engine for audio and video which could be searched based on spoken word content. Our system used speech recognition and natural language processing to create a search index of audio and video files posted publicly on the Internet. During the search process, audio and video with matching spoken words could be streamed to users. As the Vice President of Operations and Technology, I architected and helped build-out the back end of the system, which supported speech recognition, search indexing, and providing the capability for audio and video streaming in search results. Today, at RAMP Inc., the project has grown to a product that is used by media outlets such as ABC, CBS, NBC, Fox, and Reuters. In addition, during this time at BBN, I continued to be engaged with Voice-over-IP related projects through the time I left BBN.

14. Around 2008-2009, while I was Chief Technology Officer at Eons, a venture backed company founded by Jeff Taylor, who also founded the hiring web site Monster.com, Eons launched an advertising network. Eons built a network of sites on which advertisements could be placed, fulfilled client advertisement purchases, and tracked delivery of clients' advertisements. In addition, we utilized the Solr search platform in order to index the millions of items of content added by Eons members, in order to make them searchable.

15. I have also continued to develop web sites for various business projects, as well as setting up web sites on a volunteer basis for various groups that I am associated with.

16. I am the author of several publications devoted to a wide variety of technologies in the fields of electrical engineering and computer science. These publications are listed on my C.V. (App. A).

17. In summary, I have extensive familiarity with systems, networks, architectures, and methods related to traditional circuit-switched telecommunications, packet-based telecommunications, and systems that merged the two technologies, and I am familiar with what the states of these technologies were at the relevant time of the ’005 Patent invention and before.

III. Level of Ordinary Skill in the Art

18. I am familiar with the knowledge and capabilities of persons of ordinary skill in packet-based and circuit-switched telecommunication systems in the period around November 2006, the earliest claimed priority date for the ’005 Patent. I base this on my experience with VoIP starting with my streaming media work for my PhD thesis in the mid 1990’s, through my first post-degree job at NBX, which shipped the first PBX to work over a computer data network, and my later professional work in the early 2000’s developing VoIP test tools. One of ordinary skill in the art would have been familiar with VoIP gateways, which interfaced packet-based phone systems to the PSTN, softswitches, which provided call control and advanced features for VoIP phone systems, and the SIP and MGCP protocols, which provided signaling and device control for VoIP systems. One of ordinary skill in the art would also have been familiar with the types of features available in a VoIP system, such as the ability for VoIP phones to be moved around in a data network without reprogramming a switch, with softphones, which allowed users to use their computers as a telephone, and with toll bypass, where calls maybe placed out through remote gateways accessible over the data network to avoid long distance telephone charges.

19. In my opinion, the level of ordinary skill in the art that one would need in order to have the capability of understanding the scientific and engineering principles applicable to the ’005 Patent is (i) a Bachelor degree (or higher degree) in an academic area emphasizing electrical engineering and (ii) 2-4 years of industry experience in designing or developing packet-based and circuit-switched telecommunication systems. Experience and technical training may substitute for educational requirements, while advanced degrees may substitute for experience

IV. Relevant Legal Standards

20. Obviousness

a. I have been asked to provide my opinions regarding whether the claims 1, 24-26, 49-50, 73-79, 83-84, 88-89, 92, 94-96, 98, and 99 of the ’005 Patent would have been obvious to a person having ordinary skill in the art at the time of the alleged invention, in light of the prior art.

b. I have been informed and understand that a patent claim is not patentable under 35 U.S.C. § 103 if the differences between the patent claim and the prior art are such that the claimed subject matter as a whole would have been obvious at the time the claimed invention was made to a person having ordinary skill in the art to which the subject matter pertains. Obviousness, as I have been informed, is based on the scope and content of the prior art, the differences between the prior art and the claim, the level of ordinary skill in the art, and, to the extent that they exist and have an appropriate nexus to the claimed invention (as opposed to prior art features), secondary indicia of non-obviousness.

c. I have been informed that whether there are any relevant differences between the prior art and the claimed invention is to be analyzed from the view of a person of ordinary skill in the art at the time of the invention. As such, my opinions below as to a person of ordinary skill in the art are as of the time of the invention, even if not expressly stated as such; for example, even if stated in the present tense.

d. In analyzing the relevance of the differences between the claimed invention and the prior art, I have been informed that I must consider the impact, if any, of such differences on the obviousness or non-obviousness of the invention as a whole, not merely some portion of it. The person of ordinary skill faced with a problem is able to apply his or her experience and ability to solve the problem and also look to any available prior art to help solve the problem.

e. An invention is obvious if a person of ordinary skill in the art, facing the wide range of needs created by developments in the field, would have seen an obvious benefit to the solutions tried by the applicant. When there is a design need or market pressure to solve a problem and there are a finite number of identified, predictable solutions, it would be obvious to a person of ordinary skill to try the known options. If a technique has been used to improve one device, and a person of ordinary skill in the art would recognize that it would improve similar devices in the same way, using the technique would have been obvious.

f. I have been informed that a precise teaching in the prior art directed to the subject matter of the claimed invention is not needed. I have been informed that one may take into account the inferences and creative steps that a person of ordinary skill in the art would have employed in reviewing the prior art at the time of the invention. For example, if the claimed invention combined elements known in the prior art and the combination yielded results that were predictable to a person of ordinary skill in the art at the time of the invention, then this evidence would make it more likely that the claim was obvious. On the other hand, if the combination of known elements yielded unexpected or unpredictable results, or if the prior art teaches away from combining the known elements, then this evidence would make it more likely that the claim that successfully combined those elements was not obvious.

g. I have been informed that hindsight must not be used when comparing the prior art to the invention for obviousness.

h. Obviousness may also be shown by demonstrating that it would have been obvious to modify what is taught in a single piece of prior art to create the subject matter of the patent claim. Obviousness may be shown by showing that it would have been obvious to combine the teachings of more than one item of prior art. In determining whether a piece of prior art could have been combined with other prior art or combined with or modified in view of other information within the knowledge of one of ordinary skill in the art, the following are examples of approaches and rationales that may be considered:

- Combining prior art elements according to known methods to yield predictable results;
- Simple substitution of one known element for another to obtain predictable results;
- Use of a known technique to improve similar devices (methods, or products) in the same way;
- Applying a known technique to a known device (method, or product) ready for improvement to yield predictable results;
- Applying a technique or approach that would have been "obvious to try" (choosing from a finite number of identified, predictable solutions, with a reasonable expectation of success);
- Known work in one field of endeavor may prompt variations of it for use in either the same field or a different one based on design incentives or other market forces if the variations would have been predictable to one of ordinary skill in the art; or
- Some teaching, suggestion, or motivation in the prior art that would have led one of ordinary skill to modify the prior art reference or to combine prior art reference teachings to arrive at the claimed invention.

i. I have been informed that even if a *prima facie* case of obviousness is established, the final determination of obviousness must also consider "secondary considerations" if presented. In most instances, the patentee raises these secondary considerations of non-obviousness. In that context, the patentee argues an invention would not have been obvious in view of these considerations, which include: (a) commercial success of a product due to the merits of the claimed invention; (b) a long-felt, but unsatisfied need for the invention; (c) failure of others to find the solution provided by the claimed invention; (d) deliberate copying of the invention by others; (e) unexpected results achieved by the invention; (f) praise of the invention by others skilled in the art; (g) lack of independent simultaneous invention within a comparatively short space of time; (h) teaching away from the invention in the prior art.

j. I have been informed and further understand that secondary considerations evidence is only relevant if the offering party establishes a connection, or nexus, between the evidence and the claimed invention. The nexus cannot be to prior art features. The establishment of a nexus is a question of fact.

21. Claim Construction

a. I have been informed that the first step in an invalidity analysis involves construing the claims, as necessary, to determine their scope. And, second, the construed claim language is then compared to the disclosure of the prior art. In proceedings before the USPTO, I have been informed that the claims of an unexpired patent are to be given their broadest reasonable interpretation in view of the specification from the perspective of one skilled in the art. I have been informed that the ’005 Patent has not expired. In comparing the claims of the ’005 Patent to the known prior art, I have carefully considered the ’005 Patent and the ’005 Patent prosecution history based upon my experience and knowledge in the relevant field. For purposes of this proceeding I have applied the broadest reasonable interpretation of the claim terms of the ’005 Patent that is generally consistent with the terms’ ordinary and customary meaning, as one skilled in the relevant field would have understood them. Because I have been informed that the claim construction standard in this proceeding differs from that used in U.S. district court litigation, nothing herein should be taken as an indication that I consider these constructions to control in a district court setting.

b. I have been informed that a special claim construction analysis is applied to claim limitations drafted in means-plus-function format pursuant to 35 U.S.C. § 112, ¶ 6. When construing a means-plus-function limitation, I understand that the claimed function must first be identified and then the corresponding structure that actually performs the claimed function must be identified from the specification. A means-plus-function claim term is limited to the disclosed structures and equivalents.

c. I have been informed that where the disclosed structure for a means-plus-function limitation is a computer, or microprocessor, programmed to carry out an algorithm, the corresponding structure is not the general purpose computer or processor, but rather is a special purpose computer or processor programmed to perform the disclosed algorithm. In other words, I understand the disclosed algorithm is part of the corresponding structure for computer-implemented means-plus-function limitations.

d. I have been informed that where a patent specification fails to adequately disclose corresponding structure for a means-plus-function claim limitation, that claim is invalid as indefinite pursuant to 35 U.S.C. § 112, ¶ 2. But I also understand that indefiniteness may not be challenged in an *Inter Partes* Review. Accordingly, though it is my opinion that certain means-plus-function claim limitations in the ‘005 Patent Challenged Claims may not have sufficiently described corresponding structure, I have assumed they are definite and have assessed their validity based on the specific structure actually disclosed.

e. I have reviewed the proposed claim constructions in Section IV.B.2 of the Petition, agree with these proposed constructions, and have applied these constructions in my analyses herein.

V. <u>Overview of the ’005 Patent</u>

22. This overview is not meant to describe my full understanding of the ’005 Patent, but is only used to generally describe the relevant functionalities of the ’005 Patent.

23. The ’005 Patent generally describes a telephony system in which calls are classified as either public network calls or private network calls and routing messages are generated to route calls accordingly. *See* **Ex. 1001** at Abstract. A call routing controller receives a request to establish a call from a calling party, which includes an identifier of the called party. *Id.* at 1:59-61. Call routing controller then compares the called party identifier with attributes of the calling party identifier, and may reformat the called party identifier depending on the result of this comparison. *Id.* at 2:13-31. Based on the comparison of attributes of the calling party and the called party identifier, the call routing controller next determines whether the called party is a subscriber to a private network. *Id.* at 2:51-53, 3:4-8. If so, a routing message is generated so that the call can be directed to the private network node serving the called party. *Id.* at 1:64-67. If the called party is not on the private network, the call is classified as a public network call and a routing message is generated so that the call can be directed through a gateway to a public network. *Id.* at 1:67-2:2.

24. More specifically, the ’005 Patent describes a calling party utilizing a Voice over Internet Protocol (“VoIP”) telephone who is able to call (1) other VoIP subscribers on a private packet-based network or (2) standard public switched telephone network (“PSTN”) customers on the public telephone network. *Id*. at 1:20-2:2. To identify a single destination the calling party is attempting to reach, the '005 Patent teaches that modifications to the dialed digits may be necessary. Fig. 8B illustrates a variety of modifications, which include, as an example, prepending the calling party's country code and area code to the dialed digits when the called party dials a local number. *Id*. at Fig. 8B. With the formatted number, a direct-inward-dial bank ("DID") table is referenced to determine if the called party is a subscriber to the private packet network. *Id*. If not, the call is directed to a PSTN gateway and the formatted number is used to connect the call over the public PSTN to the called party. *Id*.

25. Put simply, the '005 Patent describes a system in which a VoIP subscriber may place calls to other VoIP subscribers or to phones on the PSTN. The described system also enables the calling party to enter dialed digits in a familiar manner as if the caller were placing a call on the PSTN, and the system reformats the dialed digits using caller attributes (e.g., country code, area code, etc.), determines whether the callee is on the private packet network or public PSTN, and routes the call accordingly.

VI. State of the Relevant Art in 2006

26. Circuit switched telecommunications networks, packet-switched communications networks, the Internet and the carriage of voice via IP packets over the Internet was well-known by 2006.

27. Telephonic communications dates to the 1800's, where in 1877, Alexander Graham Bell received U.S. Patent No. 186,787 titled "Improvement in Electric Telegraphy" which disclosed a device which transmits voice over a circuit, in which a bell rings when the main circuit is opened.

28. Packet communications dates to the 1960's, and was demonstrated with the creation of the ARPANET, which was funded by the Advanced Research Projects Agency of the U.S. Department of Defense. By 1983, packetized speech was carried over the ARPANET and interfaced to the PSTN in a system demonstrated by Weinstein and Forgie.[1] This system transported speech packets over the data network, and provided a packet/circuit interface between packet switches and the T1 digital carrier format of the telephone network. See Weinstein at p. 977. Around this time, Dialogic Inc. and Natural Microsystems Inc. were founded, and eventually these companies produced hardware and software for computer to telephone system integration. Computer telephony boards made by these vendors and others were used by the 1990's to create Voice-over-IP systems.

29. By the 1990's the Internet Engineering Task Force, the body tasked with developing standards for the Internet, began to propose standards that focused on real-time applications such as VoIP. "RTP: A Transport Protocol for Real-Time Applications" was released as RFC 1889 in January 1996.[2] In 1999, the IETF published "SIP: Session Initiation Protocol"[3] which is used to establish sessions, or calls, and is used frequently by VoIP systems today. Both these protocols became widely used by the mid 2000's.

30. In 1999, the global telecommunications manufacturer Siemens announced a VoIP gateway utilizing the Natural Microsystems Fusion platform[4] order to "enable[] voice and

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[1] *See* Appendix B, Weinstein and Forgie, "Experience with Speech Communications," IEEE Journal on Selected Areas in Communications, Vol. SAC-1, No. 6, December 1983.
[2] *See* Appendix C, https://tools.ietf.org/html/rfc1889 (excerpt)
[3] *See* Appendix D, https://tools.ietf.org/html/rfc2543 (excerpt)
[4] *See* Appendix E, http://www.cnet.com/news/fusion-will-help-raise-net-voices/

real-time fax calls over IP-based networks."[5]

31. In the late 1990's, business telephone systems (also known as private branch exchanges) that ran over data networks became available. NBX Corporation and Selsius Systems were among the first to release a PBX for small and medium sized businesses. Selsius was acquired by Cisco in 1998[6] and NBX was acquired by 3Com in 1999[7]. Both 3Com and Cisco were major manufacturers of packet communications equipment.

32. By 2000, companies were releasing full VoIP solutions utilizing gateways and softswitches. A softswitch is a computer running software that is able to complete calls and handle call features in a packet network that can include gateways to the PSTN. These solutions were promoted as a lower cost way to offer telecommunications services as well as lower cost long distance by providing long distance backhaul over the Internet or other data network.[8] Carrier-class gateways were available from manufacturers by the late 1990's[9] and carrier-class softswitches were available from manufacturers by the early 2000's.[10]

33. In the early 2000's, Voice-over-IP commercial phone service became available. Vonage, incorporated in 2001, was one of the early pioneers of consumer VoIP. Such a system is also known as an "over the top" system in that it is provides telephone service over a consumer's Internet connection, and thus no phone lines are required into a subscriber's

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[5] *See* Appendix F, http://pressreleases.responsesource.com/news/3014/natural-microsystems-launches-new-ip-telephony-platform/
[6] *See* Appendix G, http://www.cisco.com/chinese/warp/public/146/october98/9810p46-e.html
[7] *See* Appendix H, http://www.nytimes.com/1999/02/23/business/company-news-3com-is-acquiring-nbx-for-100-million.html
[8] *See* Appendix I, "nuVOICE – Next Generation VoIP Solutions," August 16, 2000.
[9] *See, e.g.*, Appendix J, http://www.sonus.net/resources/press-releases/sonus-networks-gateway-switch-ip-telephony-suite-are-first-capacity
[10] *See, e.g.*, Appendix K, http://www.sonus.net/resources/press-releases/sonus-networks-psx6000-softswitch-sets-standard-scalability-reliability

residence. By 2005, Vonage announced that it had one million subscribers.[11] Vonage provided features such as call forwarding, call waiting, area code selection (no matter the actual area code determined by one's physical residence[12]), and voicemail, among others.[13]

34. By November 2, 2006, the priority date of the '005 patent, softswitches (also known as call controllers), gateways between an IP network and the PSTN, the use of the Internet or a data network for saving long distance toll charges, the ability to place calls within a private network as well as to the PSTN, and the ability for users to select profile-based features such as their area code, were well-known and widely used technologies.

VII. Opinions Concerning Chu '684 and Chu '366

35. I have been asked to consider whether claims 1, 24-26, 49-50, 73-79, 83-84, 88-89, 92, 94-96, 98, and 99 are obvious over Chu '684 in view of Chu '366. It is my opinion that they are indeed obvious and that the combination of Chu '684 and Chu '366 teaches all elements of claims 1, 24-26, 49-50, 73-79, 83-84, 88-89, 92, 94-96, 98, and 99 as set forth in the claim chart for this combination in the Petition.

36. For a number of reasons, it would have been obvious to one of skill in the art to modify the system described by Chu '684 with the specific dialed digit reformatting teachings of Chu '366. First, the references are in the same technological field and substantially overlap in relevant subject matter. For example, both references teach telecommunications systems in which VoIP subscribers can place calls to a customer on the public PSTN. *Compare* **Ex. 1006**, *Chu '684* at 8:65-9:1 ("At step **608**, after receiving all the dialed digits from the phone **101**, server **110**

[11] *See* Appendix L, http://www.networkcomputing.com/networking/vonage-hits-one-million-subscribers/1486680870
[12] *See* Appendix M, https://web.archive.org/web/20051210104545/http://vonage.com/avail.php
[13] *See* Appendix N, https://web.archive.org/web/20051210104312/http://vonage.com/features.php?lid=nav_features

consults its dial plan to determine whether the call is local, to another on-net phone, or to a phone that is on the PSTN.") *with* **Ex. 1007**, *Chu '366* at 14:30-33 ("[T]here is shown a system for communications between a computing environment **202** including the application program according to the present system and a PSTN telephone **216.**"). Additionally, both references teach a process in which dialed digits and caller attributes are used to determine where the call should be routed. *Compare* **Ex. 1006**, *Chu '684* at 8:65-9:1 ("At step **608**, after receiving all the dialed digits from the phone **101**, server **110** consults its dial plan to determine whether the call is local, to another on-net phone, or to a phone that is on the PSTN.") *with* **Ex. 1007**, *Chu '366* at Fig. 6. Finally, both references expressly reference E.164 as an international standard dial plan. *Compare* **Ex. 1006**, *Chu '684* at 3:59-61 ("[E]ach IP phone [may be] assigned its own E.164 number (the international standard dial plan) and receiving calls from the PSTN directly.") *with* **Ex. 1007**, *Chu '366* at 1:18-20 ("E.164 [ ] provides a uniform means for identifying any telephone number in the world to any telephony user in the world.").

37. Second, the system of Chu '684 already contains the infrastructure needed to support such reformatting. Chu '684 expressly discloses geographically dispersed subscribers each of whom may use subscriber-specific dial plans. **Ex. 1006**, *Chu '684* at 12:60-64. ("Many subscribers, each with multiple locations, can be served by the same packet-switch/soft-switch network. Each subscriber can use their the [sic] own IP address plan as well as their own dial plan."). One of skill in the art would understand that the purpose of assigning subscriber-specific dial plans, rather than dial plans applicable to all PBX subscribers, is to include subscriber-specific information such as geographic location, area codes, etc. Thus, like as expressly disclosed in Chu '366, the infrastructure of Chu '684 would support dialed digit reformatting based on attributes of the caller such as location and area code.

38. Third, the proposed modification to Chu ’684 would be straightforward, would not require undue experimentation, and would produce predictable results. Prepending information such as country codes and area codes to dialed digits has a long history in the PSTN. Upon reading the disclosure of Chu ’684, a person of ordinary skill in the art would have recognized that allowing users to place calls as if they were dialing from a standard PSTN phone would be desirable, creating a system capable of supporting a more intuitive and user-friendly interface. One of ordinary skill would also have appreciated that these improvements to Chu ’684 could be achieved by merely programming the system of Chu ’684 to analyze the dialed digits and reformat as necessary using caller attributes such as national and area code. Such modifications are simply a combination of the system of Chu ’684 with elements of Chu ’366 that would have yielded predictable results without requiring undue experimentation. Thus, it would have been natural and an application of nothing more than ordinary skill and common sense to combine Chu ’684 with the number reformatting of Chu ’366.

39. Accordingly, it is my opinion that it would have been obvious to a person having ordinary skill in the art to modify Chu ’684 with the specific dialed digit reformatting teachings of Chu ’366.

VIII. <u>Opinions Concerning Chu ’684 and Chen</u>

40. I have been asked to consider whether claims 1, 24-26, 49-50, 73-79, 83-84, 88-89, 92, 94-96, 98, and 99 are obvious over Chu ’684 in view of Chen. It is my opinion that they are indeed obvious and that the combination of Chu ’684 and Chen teaches all elements of claims 1, 7, 27-28, 34, 54, 72-74, 92-93, and 111 as set forth in the claim chart for this combination in the Petition.

41. For a number of reasons, it would have been obvious to one of skill in the art to modify the system described by Chu ’684 with the specific dialed digit reformatting teachings of Chen. First, the references are in the same technological field and substantially overlap in relevant subject matter. For example, both references teach telecommunications systems in which VoIP subscribers can place calls to a customer on the public PSTN. *Compare* **Ex. 1006**, *Chu ’684* at 8:65-9:1 (“At step **608**, after receiving all the dialed digits from the phone **101**, server **110** consults its dial plan to determine whether the call is local, to another on-net phone, or to a phone that is on the PSTN.”) *with* **Ex. 1008**, *Chen* at Fig. 5 (Illustrating a VoIP customer SIP Phone, external number Translator, PSTN Gateway, and PSTN Switch). Additionally, both references teach a process in which dialed digits and caller attributes are used to determine where the call should be routed. *Compare* **Ex. 1003**6 *Chu ’684* at 8:65-9:1 (“At step **608**, after receiving all the dialed digits from the phone **101**, server **110** consults its dial plan to determine whether the call is local, to another on-net phone, or to a phone that is on the PSTN.”) *with* **Ex. 1008**, *Chen* at Fig. 6. Finally, both references expressly reference E.164 as an international standard dial plan. *Compare* **Ex. 1006**, *Chu ’684* at 3:59-61 (“[E]ach IP phone [may be] assigned its own E.164 number (the international standard dial plan) and receiving calls from the PSTN directly.”) *with* **Ex. 1008**, *Chen* at ¶ 006 (“E.164 is an ITU-T (International Telecommunication Union Telecommunication Standardization Sector) recommendation that defines the international public telecommunication numbering plan.”).

42. Second, the system of Chu ’684 already contains the infrastructure needed to support such reformatting. Chu ’684 expressly discloses geographically dispersed subscribers each of whom may use subscriber-specific dial plans. **Ex. 1006**, *Chu ’684* at 12:60-64. (“Many subscribers, each with multiple locations, can be served by the same packet-switch/soft-switch network. Each subscriber can use their the [sic] own IP address plan as well as their own dial plan.”). One of skill in the art would understand that the purpose of assigning subscriber-specific dial plans, rather than dial plans applicable to all PBX subscribers, is to include subscriber-specific information such as geographic location, area codes, etc. Thus, like as expressly disclosed in Chen, the infrastructure of Chu ’684 would support dialed digit reformatting based on attributes of the caller such as location and area code.

43. Third, the proposed modification to Chu ’684 would be straightforward, would not require undue experimentation, and would produce predictable results. Prepending information such as country codes and area codes to dialed digits has a long history in the PSTN. Upon reading the disclosure of Chu ’684, a person of ordinary skill in the art would have recognized that allowing users to place calls as if they were dialing from a standard PSTN phone would be desirable, creating a system capable of supporting a more intuitive and user-friendly interface. One of ordinary skill would also have appreciated that these improvements to Chu ’684 could be achieved by merely programming the system of Chu ’684 to analyze the dialed digits and reformat as necessary using caller attributes such as national and area code. Such modifications are simply a combination of the system of Chu ’684 with elements of Chen that would have yielded predictable results without requiring undue experimentation. Thus, it would have been natural and an application of nothing more than ordinary skill and common sense to combine Chu ’684 with the number reformatting of Chen.

44. Accordingly, it is my opinion that it would have been obvious to a person having ordinary skill in the art to modify Chu ’684 with the specific dialed digit reformatting teachings of *Chen*.

IX. Opinions Concerning Claim Constructions

45. I have been asked to assess whether Chu ’684 discloses using a caller identifier to locate a caller dialing profile comprising caller attributes, as claimed in the ’005 Patent

Challenged Claims. In my opinion, it does. Chu ’684 teaches that each subscriber is assigned their own dial plan, a unique IP address, and a unique E.164-compliant telephone number. **Ex. 1006**, *Chu ’684* at 3:56-64. Chu ’684 expressly notes that a subscriber’s dial plan can be determined “from the ID of the server,” but also teaches that multiple subscribers may use the same server. *Id*. at 9:30-33 and 4:25-28. Accordingly, one of skill in the art would understand that the system described by Chu ’684 must necessarily use unique subscriber-specific information in addition to the server ID to identify the caller’s dial plan. Such subscriber-specific information would be the subscriber’s E.164-compliant telephone number, globally unique database key, or the like. In sum, Chu ’684 teaches, expressly or inherently, locating a caller dialing profile using a caller identifier as required by the Challenged Claims of the ’005 Patent.

46. I have been asked to assess whether the prior art combinations above teach the same or equivalent structures as certain means-plus-function limitations in claims 50 and 73 of the ’005 Patent. As detailed in the following analyses, I have concluded that each prior art combination does teach these structures or their equivalents.

47. Claim 50 recites a “means for using a caller identifier associated with the caller to locate a caller dialing profile comprising a plurality of calling attributes associated with the caller.” Although I disagree that the ’005 Patent provides sufficient detail regarding the specific structure responsible for locating a caller dialing profile, I understand indefiniteness cannot be challenged in an IPR. To the extent any corresponding structure is disclosed that performs this recited function, it is (at least under the broadest reasonable interpretation) RC processor circuit 200 programmed to implement the algorithm illustrated in cell 254 of Fig. 8A. **Ex. 1001**, *’005 Patent* at 19:38-45, FIGS. 7, 8A. Cell 254 states that the processor “[u]se(s) caller field to get dialing profile for caller from database.” *Id.* at FIG. 8A. In other words, the corresponding structure is a processor programmed to use some information about the caller to retrieve that caller's dialing profile. Given the lack of detail in the corresponding structure for this limitation, there is no question that Chu '684, Chu '366, and Chen are all processor-based systems that necessarily teach processors for locating caller dialing profiles that are at least equivalent to the disclosed RC processor circuit 200 programmed to perform this algorithm. *See, e.g.,* **Ex. 1006**, *Chu '684* at 3:56-64, 4:59-63, 9:30-33, 12:60-66 (detailing soft-switch locating caller dial plan); **Ex. 1007**, *Chu '366* at 2:9-15 (call origin location profile established by the caller that is necessarily located by a processor); **Ex. 1008**, *Chen* at ¶ 0033, Fig. 6 (detailing caller's dial plan that is necessarily located by a processor).

48. Claim 50 recites the following limitations:

- "means for, when at least one of said calling attributes and at least a portion of a callee identifier associated with the callee meet private network classification criteria, producing a private network routing message for receipt by a call controller, said private network routing message identifying an address, on the private network, associated with the callee"
- "means for, when at least one of said calling attributes and at least a portion of said callee identifier meet a public network classification criterion, producing a public network routing message for receipt by the call controller, said public network routing message identifying a gateway to the public network"

Each of these means-plus function limitations corresponds to a computer-implemented algorithm disclosed in the '005 Patent. Accordingly, I understand the corresponding structure I am to assess for purposes of my validity analysis must include these specifically disclosed algorithms. Specifically, the portions of these limitations directed to matching a callee identifier and caller attributes to classify a call as private or public describe aspects of the algorithm illustrated in Fig. 8B and described in the corresponding sections of the '005 Patent specification. Fig. 8B, reproduced below, illustrates various alternate branches by which (1) caller attributes are compared to a callee identifier (e.g., dialed digits), (2) the callee identifier is reformatted using caller attributes, and (3) the call is classified as either public or private using the reformatted callee identifier.

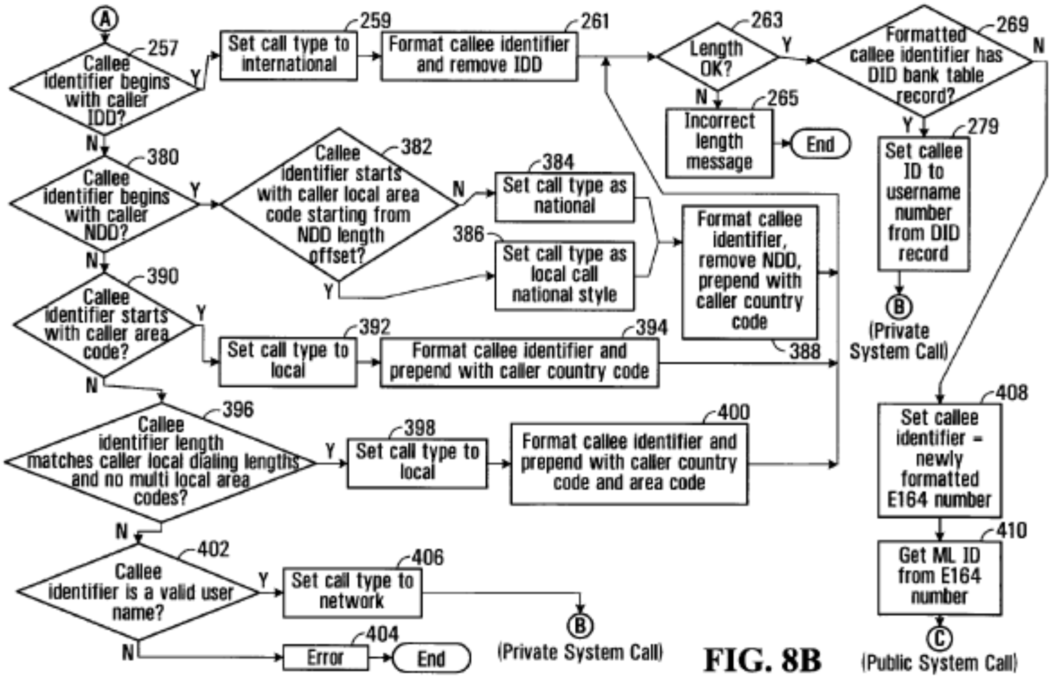
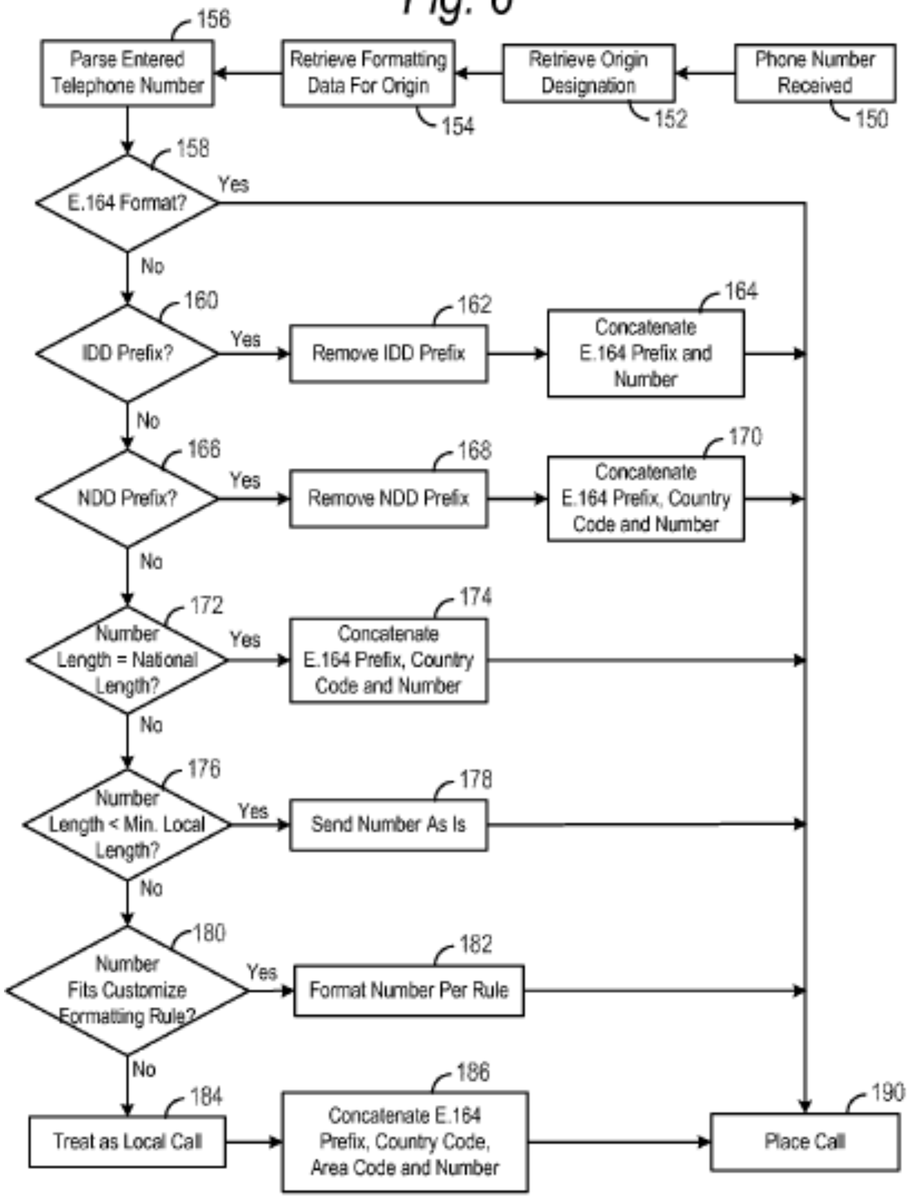


FIG. 8B

For example, if the caller dials as short-form, seven digit local call (e.g., 123-4567), the algorithm will proceed through (1) block 257, confirming there is no IDD matching the caller's IDD, (2) block 380, confirming there is no NDD matching the caller's NDD, (3) block 390, confirming there is no area code that matches the caller's, (4) block 396, confirming the dialed digit length matches the caller's local dialing length, (5) block 398, setting the call type to "local," (6) block 400, prepending the caller's country code and area code to the dialed digits, and (7) blocks 263 and 269, determining whether the reformatted callee identifier corresponds to a user on the private network or to a user on the public network. As described above in Sections VII and VIII, the combinations of (1) Chu '684 and Chu '366 and (2) Chu '684 and Chen teach this very process by which dialed digits are compared to caller attributes, modified appropriately, and then the reformatted callee identifier is used to determine whether the call is a private VoIP call or a public PSTN call. In fact, Chu ’366 and Chen—both relied upon for their comparison and dialed digit reformatting teachings—disclose identical algorithms to the above-described algorithm in Fig. 8B of the ‘005 Patent. For example, like the ‘005 Patent algorithm, Fig. 6 in Chu ’366, reproduced below, teaches (1) checking for an IDD prefix, (2) checking for an NDD prefix, (3) checking the local dialing length, and (4) prepending the caller’s country code and area code to the dialed digits:

Fig. 6



Similarly, Fig. 6 in Chen, reproduced below, teaches (1) checking for an IDD prefix, (2) checking for an NDD prefix, (3) checking the dialed digit length, and (4) prepending the caller's country code and area code to the dialed digits:

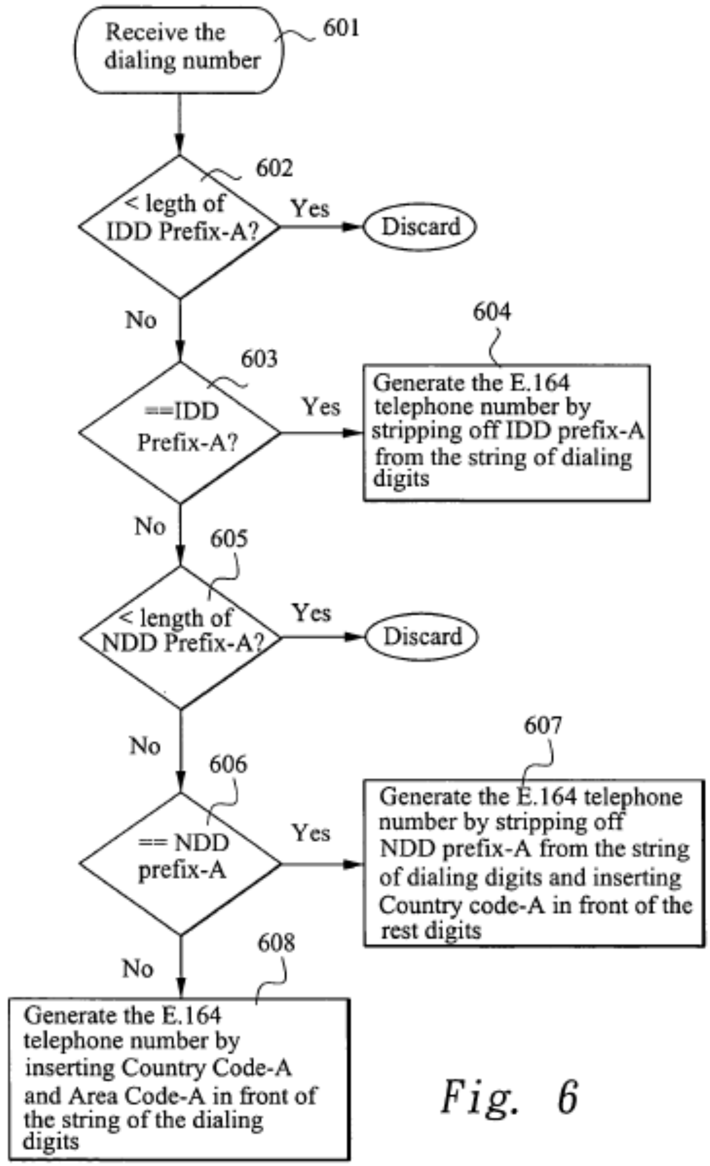


Fig. 6

As illustrated in the above figures, both Chu ’366 and Chen teach algorithms for reformatting the callee identifier using caller attributes that are nearly identical, and certainly equivalent, to the algorithm disclosed in Fig. 8B of the ‘005 Patent. Additionally, Chu ’684 teaches analyzing a callee identifier to determine whether the call should be directed to a subscriber on the private packet network or to a customer on the public PSTN. Accordingly, it is my opinion that the combinations of (1) Chu ’684 and Chu ’366 and (2) Chu ’684 and Chen teach the same or equivalent structure to claim 50 with respect to the “matching” functions. With respect to producing network messages, the corresponding structure is processor 202 of RC processor circuit 200, programmed to implement the algorithms illustrated in cell 350 of Fig. 8A or cell 644 of Fig. 8C for private messages and the claimed function for public messages. **Ex. 1001**, *’005 Patent* at 20:37-58, 24:55-25:12, 26:49-57, FIGS. 7, 8A, 8C, 8D, 16, 25, 32. For producing private network messages, cell 350 in Fig. 8A, in relevant part, states that the private network routing message should include the “IP address or domain” of the callee. *Id*. at FIG. 8A. Similarly, although less detailed regarding the specific address associated with the callee, cell 644 of Fig. 8C states that the “address of current node” is stored in the routing message buffer. As illustrated in the sample private network routing messages illustrated in Figs. 16 and 32, the address of the supernode associated with the callee is at least the domain name of that supernode, i.e., an address identifying the call controller associated with the callee. For producing public network routing messages, Fig. 8D details a process by which the address of a gateway that enables connection to the callee is selected. *Id*. at FIG. 8D. Fig. 8D illustrates additional detail regarding selecting the optimal gateway from a plurality of viable options, but, at least under the broadest reasonable interpretation applicable here, it is my opinion that these additional details should not be considered part of the associated structure for the recited function. Setting aside the process for selecting an optimal gateway, the disclosed algorithm is nothing more than obtaining an IP address of an egress gateway. Given the nature of the disclosed algorithms, which (at least under the broadest reasonable interpretation) require only that the IP address or domain corresponding to an IP-based callee or the IP address for a gateway providing access to the public network on which the callee resides, there is no question that Chu ’684 includes a processor that is at least equivalent to the disclosed processor 202 of RC processor circuit 200, programmed to implement the algorithms illustrated in cell 350 of FIG. 8A or cell 644 of Fig. 8C for producing private network routing messages and the recited function for public network routing messages. *See, e.g.,* **Ex. 1006**, *Chu '684* at 9:30-49 (detailing processor-based soft-switch producing routing messages including an IP address corresponding to the called party's egress packet switch), 13:12-34 (detailing processor-based soft-switch producing routing messages including an IP address of a gateway to the PSTN associated with the called party).

49. Finally, claim 73 recites "means for causing the private network routing message or the public network routing message to be communicated to a call controller to effect routing of the call." Although I do not concede that the '005 Patent provides sufficient detail regarding the specific structure responsible for causing routing messages to be communicated, I understand indefiniteness cannot be challenged in an IPR. To the extent any corresponding structure is disclosed that performs this recited function, it is (at least under the broadest reasonable interpretation) processor 202 of RC processor circuit 200, programmed to perform the algorithms illustrated in cell 381 of FIG. 8A, 646 of FIG. 8C, and cell 568 of FIG. 8D. **Ex. 1001**, *'005 Patent* at 20:37-58, 24:55-25:12, 26:52-53, FIGS. 7, 8A, 8C, 8D. Both these algorithms are nothing more than the processor forwarding routing messages to a call controller. Given the lack of detail in the corresponding structure for this limitation, there is no question that Chu '684 includes a soft-switch that is at least equivalent to the disclosed processor 202 of RC processor circuit 200, programmed to perform the algorithms illustrated in cell 381 of FIG. 8A, cell 646 of FIG. 8C, and cell 568 of FIG. 8D. *See, e.g.,* **Ex. 1006**, *Chu '684* at 4:52-56, 9:30-49, 13:12-34 (detailing soft-switch producing and communicating routing messages).

X. <u>Opinions Concerning Claims 25, 79, 84, 89, and 92</u>

50. I have been asked to consider whether the combinations of (1) Chu ’684 and Chu ’366 and (2) Chu ’684 and Chen teach Claim 25, which recites “A non-transitory computer readable medium encoded with codes for directing a processor to execute the method of claim 1.” In my opinion, both combinations satisfy this limitation, which is nothing more than an additional requirement that the steps of Claim 1 are implemented in software. As discussed at length above, Chu ’684, Chu ’366, and Chen are processor-based systems that necessarily implement their functions in software, i.e., with codes for directing a processor. One of skill in the art would understand this fact and would recognize that both combinations necessarily satisfy the additional limitation recited in Claim 25.

51. I have been asked to consider whether the combinations of (1) Chu ’684 and Chu ’366 and (2) Chu ’684 and Chen teach Claims 79 and 84, which recite, respectively, “The method of claim 74, wherein the packet switched network is accessed via an Internet service provider” and “The method of claim 74, wherein the address in the first portion is accessible through the first participant's Internet service provider.” In my opinion, both combinations satisfy these claims. Chu ’684 teaches that subscribers “can use . . . “public internet [sic] addressing,” which necessarily means that the subscriber accesses the packet network via an Internet service provider. **Ex. 1006**, *Chu ’684* at 13:4-9. One of skill in the art would understand this fact and would recognize that, based on this unambiguous teaching in Chu ’684, both combinations necessarily satisfy the additional limitations recited in Claims 79 and 84.

52. I have been asked to consider whether the combinations of (1) Chu ’684 and Chu ’366 and (2) Chu ’684 and Chen teach Claims 89 and 92, which recite, respectively, “The method of claim 74, wherein the second network classification criterion is satisfied when access to the second participant requires routing through a portion of the packet switched network operated by a communication service supplier" and "The method of claim 74, wherein the address in the second portion of the packet switched network comprises an address accessed by a communication service supplier." In my opinion, both combinations satisfy this limitation. Chu '684 teaches Chu '684 teaches a soft-switch entity that generates a private network routing message identifying an address on the second portion of the network. **Ex. 1003**, *Chu '684* at 13:66-14:21. Chu '684 further teaches that routing messages for inter-VPN calls may be sent from one soft-switch to another peer soft-switch. *Id.* at 4:52-56. One of skill in the art would understand that the second portion of the network is necessarily operated by a communications service supplier and that the call is classified as an inter-VPN call whenever the callee is identified as residing on the second network that is operated by the communications service supplier. Accordingly, one of skill in the art would conclude that both combinations (each based on Chu '684) necessarily satisfy the additional limitations recited in Claims 89 and 92.

I declare that all statements made herein of my knowledge are true, and that all statements made on information and belief are believed to be true, and that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code.

Date: 6/14/16

By: Henry H. Houh
Henry H. Houh

APPENDIX A

Henry H. Houh

Education

Massachusetts Institute of Technology, Cambridge, MA

- PhD in Electrical Engineering and Computer Science, January 1998. "Designing Networks for Tomorrow's Traffic," thesis supervised by Professor David Tennenhouse and Professor John Guttag. GPA 4.7/5.0
- Master of Science in Electrical Engineering and Computer Science, February 1991. "Demonstration of a laser repetition rate multiplier," thesis. GPA 4.5/5.0
- Bachelor of Science in Electrical Engineering and Computer Science, June 1989. "Boundary element analysis of arbitrarily shaped dielectric structures," thesis. GPA 4.7/5.0
- Bachelor of Science in Physics, February 1990. GPA 4.7/5.0

Experience

H3XL Inc. d/b/a Einstein's Workshop (formerly Lexington Robotics)

- 2009 - present: Founder and President. Started local league providing science and engineering education programs based on LEGO Mindstorms, LEGO WeDo, and FIRST LEGO League. In 2012, grew program into a full science, technology, engineering and math enrichment program and creative/maker space, in 7,000 square feet of space. Serve 2,000+ kids and families annually. As of end of 2015, have delivered an estimated 65,000 student-instruction-hours of STEM courses. Principle Investigator for 2-year DARPA grant to improve 3D Computer Aided Design tool developed in-house for the purposes of teaching 8 year olds and up 3D CAD.

Houh Consulting Inc. / Independent Consultant

- 2009 - present: Technical consultant specializing in Social Networking, Web 2.0, Web Site Development, Data Networking, Optical Networking, Telecommunications, Media Streaming and Voice Over IP. Representative clients include: BBN, Akin Gump Strauss Hauer & Feld LLP, Covington & Burling LLP, Winston & Strawn LLP, Wilmer Cutler Pickering Hale and Dorr LLP, Kellogg Huber Hanson Todd Evans & Figel PLLC, McGuireWoods LLP, and Sidley Austin LLP.

Eons

- 2008 - 2009: Chief Technology Officer. Created product that Eons acquired from BBN Technologies. Integrated BBN product with Eons social networking platform and significantly increased the Eons group creation rate. Helped evaluate advertisement platform offerings and rolled out the "Boom Network" advertising network. Eons raised $32 million from General Catalyst Partners, Charles River Ventures, Sequoia Capital, and Intel Capital.

BBN Technologies

- 2007 - 2008: Delta Division, Vice President of Technology. Grew “Boomerang” counter-sniper project engineering team and significantly de-risked $10 million worth of product deliveries. Boomerang was a significant asset leading to the acquisition of BBN by Raytheon in 2009. Created new business plan and grew team; launched new fully-featured social networking web site in 5 months. Served as lead expert witness in patent infringement lawsuit, resulting in $58 million jury award to client; verdict for patents I testified on were upheld on appeal which resulted in a $120 million settlement.
- 2004 - 2007: Delta Division, Director of Technology, responsible for commercializing IP and creating new businesses. Hired and grew division's initial engineering team. Wrote three business plans, two of which are funded and active. For call center business plan, acted as general manager, hiring and managing engineering team, inside sales team, and identifying and recruiting a new general manager. Identified and recruited other key employees to Delta Division, including senior members of team leading to successful internal sales growth and spin-outs of projects. Contracted by BBN to BBN spin-out PodZinger as VP of Operations and Technology. Identified sales team for counter-sniper system, leading to $10 million dollars of new sales within 6 months and $100+ million in additional orders in the following two years.

PodZinger Inc. (BBN spin-out, also known as EveryZing and now RAMP)

- 2006: Vice President of Operations and Technology. Significantly upgraded capability of consumer-facing search site and redeployed web site from company to co-location facility. Identified key portions of infrastructure for upgrading and cost reduction. Helped write business plan, evaluating advertisement models of revenue. Hired in operations replacement and phased back to BBN.

Commonwealth Capital

- 2004: Entrepreneur-in-residence (informally), performed technical due diligence on business plans, brainstormed ideas for new businesses with venture partner. With venture partner, left for portfolio company BBN to form core of commercialization team.

Empirix, Inc./Teradyne, Inc.

- 2001 - 2004: Chief Technologist, Engineering Manager, Web Application Test Group. Researched potential new product areas; developed product plan and prototype. Responsible for three new and existing products. Managed off-shore development team. Chief architect for all web testing products. Re-architected core testing product, helped write javascript interpreter. Provided technical vision for core product.
- 2000 - 2001: Chief Technologist, Communication Infrastructure Test Group. Responsible for incorporating new technology internally, tracking new technologies, technical evaluation of partnerships and potential acquisitions. Helped develop division strategy. Developed plans which formed core capabilities for successful new products introduced in 2004-5.
- 2000 - 2001: Engineering Manager, Communications Infrastructure Test Group. Execution of new product plan developed in prior role. Grew team from four existing engineers to team of over 30 on immediate team and over 40 on project. Delivered new platform in one year. Platform

and derivatives accounted for large portion of booked products for the division within 2 years and is currently (2008) a key portion of new product offerings.

- 2000: Empirix was formed as a spin-out of Teradyne in January 2000. Reported to CEO in carve-out of Empirix from Teradyne.
- 1999: (Teradyne) Director of Business Development, Software Test Units. Reported directly to Chairman of the Board/Founder and then to general manager of software test unit (6 divisions of Teradyne). Evaluated and researched acquisition and partnership candidates. Internally assessed technology position in market and gaps in product lines. Worked with senior division staff to develop new product strategies. Attended internal Teradyne divisional board meetings. Chairman served as my mentor.

3Com Corporation/NBX Corporation

- 1999: Software Engineer 5. Continued work after 3Com acquisition of NBX. Built cross-division relationships for new products and research directions. NBX was acquired by 3Com in March 1999.
- 1997 - 1999: (NBX) Senior Scientist and Engineer. Work in IP Telephony. Architected next-generation product. Protocol design and validation for core protocol now used tens of millions of times daily. Led team in integration of IP protocols into current product. Designed audio reconstruction algorithms. Developed applications for bug analysis and diagnosis of system problems. Implementation of network simulator. Work on collaborative projects with external partners. Worked to identify gaps in product. Representative at numerous trade shows. Innovated novel methods of using product.

MIT Department of Electrical Engineering and Computer Science, Cambridge, MA

- 1991 - 1998: Research assistant, Telemedia Network Systems Group. Design, development, and implementation of Gigabit ATM network for distributed multimedia system. Studied host interface design issues. ATM network simulation.
- Spring 1989, Fall 1990, Spring 1995: Teaching assistant, Computation Structures digital systems course. (Spring 1995 Head TA)
- 1988 - 1989: Head laboratory teaching assistant for Computation Structures. Responsibilities included writing and revising lab assignments, and maintaining the lab.
- 1987: Laboratory teaching assistant for Computation Structures.
- 1987: Design, construction, and programming of 16-bit computer.

Agora Technology Group, Incorporated

- 1994 - 1996: Founder and CEO. Conceived and oversaw development of targeted advertising-supported Web sites. Responsible for company's vision and direction. Attended the first two WWW Conferences, presenting a workshop and paper at the first, and appearing on the “Commercialization and Economics of the Web” panel and chairing the “Where Commercial Services and the Web Are Headed” panel at the second. Sold company intact; is currently an operating stand-alone company.

AT&T Bell Laboratories, Holmdel, NJ

- 1989 - 1990: Implementation of cascadable all-optical fiber logic gate. Modelocking of all-fiber erbium laser. Construction of modelocked laser repetition rate booster. Strong optics laboratory and fiber optic experience.
- Summer 1988, 1987: Research in integrated optics. Analysis of rectangular waveguides using microwave modeling. Fabrication of integrated optical components.

Honors

- MIT Alumni Association Great Dome Award, 2010, Baker 60th Anniversary Reunion Co-Chair (highest group award given by MIT Alumni Association)
- MIT Alumni Association Presidential Citation Award (now known as Great Dome), 2008, Member of MIT Chairman’s Salon committee
- MIT Alumni Association Bronze Beaver Award, 2007 (highest individual award given by MIT Alumni Association)
- MIT Alumni Association Volunteer Honor Roll, February 2004
- MIT Alumni Association Lobdell Award, 1999
- Boston Museum of Science Gold Pin for 1000 hours of Volunteer Service, April 1999
- MIT Alumni Association Presidential Citation Award (now known as Great Dome), 1997, Member of Alumni Online Communications Committee

Patents and Patent Publications

- US Patent #7,975,296, L. Apfelbaum, H. Houh, T. Mayberry and G. Friedman, “Automated security threat testing of web pages,” July 5, 2011. See also US20030159063, WO2003067405.
- US Patent #7,877,736, H. Houh and J. N. Stern, “Computer language interpretation and optimization for server testing,” January 25, 2011. See also US20050138104, WO2005043300.
- US Patent #7,801,910, H. Houh and J. N. Stern, “Method and apparatus for timed tagging of media content,” September 21, 2010. See also US20070112837, US20090222442, WO2007056535.
- US Patent #7,590,542, D. C. Williams, W. C. Hand, H. Houh, A. R. Seeley, “Method of Generating Test Scripts Using a Voice-Capable Markup Language,” September 15, 2009. See also EP1530869, US20030212561, WO2003096663.
- US Patent #6,967,963, H. H. Houh, P. Anderson, C. Gadda, “Telecommunication method for ensuring on-time delivery of packets containing time-sensitive data,” November 22, 2005. See also EP1060400, WO2000033092, CA2318774.
- US Patent #5,144,375, M. C. Gabriel, H. H. Houh, N. A. Whitaker, “Sagnac Optical Logic Gate,” September 1, 1992. Also issued as European Patent # EP0456422, July 23, 1997, German Patent #DE69126913, August 28, 1997
- US Patent Application Publication No. 20020015387, “Voice Traffic Packet Capture and Analysis Tool for a Data Network” (Abandoned in 2007)
- US Patent Application Publication No. 20020016708, “Method and Apparatus for Utilizing a Network Processor as Part of a Test System” (Abandoned in 2007)
- US Patent Application Publication No. 20020016937, “Method and Apparatus for Utilizing a Network Processor as Part of a Test System.” (Abandoned in 2007) See also WO2002011413.

- US Patent Application Publication No. 20070106646, “User-directed navigation of multimedia search results” (Abandoned in 2009)
- US Patent Application Publication No. 20070106660, “Method and apparatus for using confidence scores of enhanced metadata in search-driven media applications” (Abandoned in 2009)
- US Patent Application Publication No. 20070106685, “Method and apparatus for updating speech recognition databases and reindexing audio and video content using the same” (Pending as of 2/24/2015)
- US Patent Application Publication No. 20070106693, “Methods and apparatus for providing virtual media channels based on media search” (Pending as of 2/24/2015)
- US Patent Application Publication No. 20070106760, “Methods and apparatus for dynamic presentation of advertising, factual, and informational content using enhanced metadata in search-driven media applications.” (Pending as of 2/24/2015) See also WO2007056485.
- US Patent Application Publication No. 20070118873, “Methods and apparatus for merging media content” (Pending as of 2/24/2015)
- US Patent Application Publication No. 20090222442, “User-directed navigation of multimedia search results” (Pending as of 2/24/2015)
- US Patent Application 11/395,732, “Search snippet creation for audio and video data” (Abandoned)
- US Patent Application 11/774,931, “Methods and apparatus for managing a social networking web site”
- US Patent Application 11/774,947, “Methods and apparatus for organizing media files”
- US Patent Application 11/774,956, “Methods and apparatus for managing an online event”
- US Provisional Patent Application 61/086,909, “Measuring and ranking relationship activity”
- US Provisional Patent Application 61/086,914, “Detecting media object commonality”
- US Provisional Patent Application 61/086,904, “Message categorization based on message characteristics”
- US Provisional Patent Application 61/086,905, “Photo tagging to request action”

Trials and Depositions

- Case No. 1:06CV682 (CMH/BRP), Verizon vs. Vonage, US District Court for the Eastern District of Virginia, filed expert report, was deposed and testified at trial.
- Case No. 1:08CV157 (CMH/TRJ), Verizon vs. Cox, US District Court for the Eastern District of Virginia, was deposed and testified at trial.
- Case No. 5:09-cv-476, Two-Way Media vs. AT&T, US District Court for the Western District of Texas, filed expert report, testified at trial
- Case No. 2:10-cv-248 (RAJ/FBS), ActiveVideo Networks vs. Verizon, US District Court for the Eastern District of Virginia, filed expert report and was deposed as an expert witness
- Case No. 1:11-cv-00880-TSE-JFA, Bear Creek Technologies, Inc. vs. Verizon Services Corp., et al, US District Court for the Easter District of Virginia, was deposed as an expert witness
- Case No. 3:10-CV-298-BBC, AlmondNet, Inc. vs. Microsoft Corp., US District Court for the Western District of Wisconsin, filed expert report
- Case No. 6:10-cv-00597, Guardian Media Technologies, Ltd. Vs. AT&T Operations, Inc. et al., US District Court for the Eastern District of Texas, Tyler Division, filed expert report
- Case No. ESCV2010-02282C, The Octopus Solution LLC v. Gary Brown et al., Essex, MA Superior Court, testified at trial

- Investigation No. 337-TA-882, In the matter of Certain digital media devices, including televisions, Blu-ray disc players, home theater systems, tablets and mobile phones, components thereof and associated software, U.S. International Trade Commission, filed expert reports, was deposed and testified at hearing
- Investigation No. 337-TA-995, In the matter of Certain communications or computing devices, and components thereof, U.S. International Trade Commission, filed expert reports, and was deposed
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- Case IPR 2014-00039, Microsoft Corporation v. B.E. Technology LLC, U.S. Patent No. 6,628,314. Submitted declaration and cross-examined in deposition.
- Case IPR 2014-00086, Apple, Inc. v. Evolutionary Intelligence, LLC, U.S. Patent No. 7,010,536. Submitted declaration and cross-examined in deposition.
- Case No. 6:11-CV-421, Stragent, LLC v. Intel Corporation, US District Court for the Eastern District of Texas, Tyler Division, deposed and testified at trial as a fact witness.

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Leadership, Activities and Interests

- Leadership
  - Discovery Museums (Acton, MA)
    - Science and Technology Advisory Council, 2012 - present
  - MIT Alumni Association Board of Directors
    - K-12 STEM Initiatives Co-chair, 2013 - present
    - Awards Committee Chair, 2012 – 2014
    - Awards Committee, 3 year term, 2011 - 2014
    - Vice President, 2 year term, 2004 - 2006
    - Board Member, 2 year term, 1997 - 1999
  - MIT Club of Boston
    - Board of Directors, 2006 - 2011
    - K-12 Initiatives Chair, 2009 - 2012
    - VP of Communications, MIT Club of Boston, 2003 - 2006
    - Past-President, MIT Club of Boston, 2002 - 2003
    - President, MIT Club of Boston, 2001 - 2002

  - President-Elect, MIT Club of Boston, 2000 - 2001
  - VP of Programs, MIT Club of Boston, 1999 - 2000
  - Activities Super-Chair, MIT Club of Boston, 1998 - 1999
- MIT Enterprise Forum of Cambridge, Inc.
  - Past Chair, 2009 – 2011
  - In-NOW-vation Co-chair, 2010
  - Chair, 2007 - 2009
  - Vice Chair, 2005 - 2007
  - Executive Board Member, 2002 - 2011
  - Winter Workshop Co-Chair, February 2007 - conceived idea for conference, which sold-out and produced largest attendance numbers in recent memory
  - Spring Workshop Co-Chair, Spring 2004
  - Membership Committee Chair, Fall 2003 - 2006
  - 25th Anniversary Dinner Chair, Fall 2003
  - As Membership Chair and Board Member, started Special Interest Groups in 2004; a SIG won the MIT Presidential Citation award, the MIT Alumni Association's highest award for organizations, in 2006
- Estabrook Elementary School PTA
  - Advisory committee to the superintendent on PCB issue, 2010-2011
  - 4th Grade after-school science program co-organizer, 2010-2012
  - 4th and 5th Grade before-school Math Olympiad co-organizer, 2009-2013
  - 5th Grade BBQ and Yearbook Committee, 2011, 2013
  - Family Math Night volunteer, 2008-2012
- Tau Beta Pi National Engineering Honor Society
  - Advisor, MA B Chapter at MIT, 2003 - present
  - District Director (National Officer), Tau Beta Pi, New England Area, 1991 - 2003
  - President, MA B Chapter at MIT, Fall 1988 - Spring 1989
  - Laureate award, 1989
- MIT Class of 1989
  - Secretary, five consecutive 5 year terms, 1989 - 2014
  - 20-year Reunion Committee and Gift Committee, 2009
  - 15-year Reunion Committee and Gift Committee, 2004
  - 10-year Reunion Committee and Gift Committee, 1999
  - 5-year Reunion Committee, 1994
  - Interim Treasurer, 1993 - 1994
  - Instituted annual senior class career fair, now entering eleventh year, now raising over $100,000 annually for senior class activities, Fall 1988
- Strong, consistent record of leadership dating to high school

- Acting
  - ‘21’ (Sony Pictures), credited as “Chinatown Dealer,” 2007, Kevin Spacey's movie about the MIT Blackjack Team inspired by “Bringing Down the House” by Ben Mezrich, opened nationwide on March 28, 2008. 21 was the number one movie in US for two weeks and number one globally for one week. 21 also topped the DVD sales, Blu-ray sales and DVD rental charts.
  - Spring Lake Theater Company, first New York-area off-broadway production of “A Chorus Line,” played role of Mark, Summer 1990
- Former member of the MIT Blackjack team

- Producer for 10,000 Maniacs’ 2013 album “Music from the Motion Picture”
- Executive Producer for 10,000 Maniacs’ 2015 album “Twice Told Tales”
- Violist, violist, harpist, guitarist, singer, actor: played in many amateur/semi-professional groups including Merrimack Valley Philharmonic, Longwood Symphony, MIT Symphony, MIT Summer Philharmonic Orchestra and Somerville Community Chorus

APPENDIX B

Experience with Speech Communication in Packet Networks

CLIFFORD J. WEINSTEIN, MEMBER, IEEE, AND JAMES W. FORGIE, MEMBER, IEEE

Abstract—The integration of digital voice with data in a common packet-switched network system offers a number of potential benefits, including reduced systems cost through sharing of switching and transmission resources, flexible internetworking among systems utilizing different transmission media, and enhanced services for users requiring access to both voice and data communications. Issues which it has been necessary to address in order to realize these benefits include reconstitution of speech from packets arriving at nonuniform intervals, maximization of packet speech multiplexing efficiency, and determination of the implementation requirements for terminals and switching in a large-scale packet voice/data system. A series of packet speech systems experiments to address these issues has been conducted under the sponsorship of the Defense Advanced Research Projects Agency (DARPA).

In the initial experiments on the ARPANET, the basic feasibility of speech communication on a store-and-forward packet network was demonstrated. Techniques were developed for reconstitution of speech from packets, and protocols were developed for call setup and for speech transport. Later speech experiments utilizing the Atlantic packet satellite network (SATNET) led to the development of techniques for efficient voice conferencing in a broadcast environment, and for internetting speech between a store-and-forward net (ARPANET) and a broadcast net (SATNET). Large-scale packet speech multiplexing experiments could not be carried out on ARPANET or SATNET where the network link capacities severely restrict the number of speech users that can be accommodated. However, experiments are currently being carried out using a wide-band satellite-based packet system designed to accommodate a sufficient number of simultaneous users to support realistic experiments in efficient statistical multiplexing. Key developments to date associated with the wide-band experiments have been 1) techniques for internetting via voice/data gateways from a variety of local access networks (packet cable, packet radio, and circuit-switched) to a long-haul broadcast satellite network and 2) compact implementations of packet voice terminals with full protocol and voice capabilities.

Basic concepts and issues associated with packet speech systems are described. Requirements and techniques for speech processing, voice protocols, packetization and reconstitution, conferencing, and multiplexing are discussed in the context of a generic packet speech system configuration. Specific experimental configurations and key packet speech results on the ARPANET, SATNET, and wide-band system are reviewed.

I. Introduction

PACKET techniques provide powerful mechanisms for the sharing of communication resources among users with time-varying demands, and have come into wide use for provision of data communications services to the military and commercial communities. The primary application of packet techniques has been for digital data communications where the bursty nature of user traffic can be exploited to achieve large efficiency advantages in utilization of communication resources. Packet networks [1]–[8] using a variety of point-to-point and broadcast transmission media have been developed for these applications, and techniques have been developed for internetwork communication [10], [11] among dissimilar nets.

Packet techniques offer significant benefits for voice as well as for data [15]–[33]. The integration of digital voice with data in a common packet-switched system offers potential cost savings through sharing of switching and transmission resources [30], as well as enhanced services for users who require access to both voice and data communications [59]–[61]. Packet internetworking techniques can be applied to provide intercommunication among voice users on different types of networks. Significant channel capacity savings for packet voice can be achieved by transmitting packets only when speakers are actually talking (i.e., during talkspurts). The silence intervals can be utilized for other voice traffic or for data traffic. Packet networks offer significant advantages for digital voice conferencing in terms of channel utilization (only one of the conferees needs to use channel capacity at any given time) and in terms of control flexibility. A packet network allows convenient accommodation of voice terminals with different bit rates and data formats. Each voice encoder will use only the channel capacity necessary to transmit its information rather than the fixed minimum bandwidth increment typically used in circuit-switched networks. The digitization of voice in packet systems provides the opportunity for security techniques to be applied as necessary to the speech traffic. Secure packet data communication techniques [13] can be applied as well for data users who require this service. Packet networks also provide a system environment for effective exploitation of variable-bit-rate voice transmission techniques, either to reduce average end-to-end bit rate or to dynamically adapt voice bit rate to network conditions.

It has been necessary to address a number of issues in order to develop the techniques required to realize these benefits. The development of packet protocols for call setup and speech transport, and strategies for reconstitution of speech from packets arriving at nonuniform intervals have been required. Other issues include the development of efficient packet speech multiplexing techniques,

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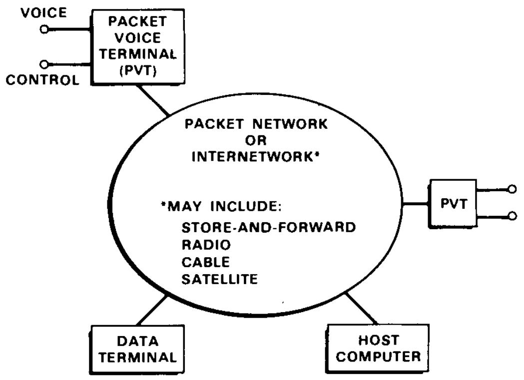

Fig. 1. Generic packet speech system configuration.

and the minimization of packet overhead and effective traffic control strategies to allow network links to be heavily loaded without saturation. System developments have been undertaken to help assess the implementation requirements for terminals and switching in a large-scale packet voice/data system, and efforts continue to drive down the size and cost of system components.

A series of packet speech experiments and system developments to address these issues has been conducted under the sponsorship of the Defense Advanced Research Projects Agency (DARPA). These efforts were initiated in 1973 by Dr. R. E. Kahn of the DARPA Information Processing Techniques Office (IPTO), who has provided leadership and numerous technical contributions through the course of the work. As will be noted in this paper and in the references, numerous individuals in several organizations have made significant contributions to the system development and experiments. The purpose of this paper is to review and evaluate the experience gained so far from these efforts in packet speech systems experiments. The perspectives and conclusions are the responsibilities of the authors and are necessarily influenced by the specific involvement of ourselves and our colleagues at Lincoln Laboratory.

This paper will begin by describing basic concepts and issues associated with packet speech systems. A generic packet speech system configuration will be described, and requirements and techniques for digital speech processing, protocol functions, packetization and reconstitution, conferencing, and multiplexing will be discussed. With this as a point of reference, the experimental system configurations and key results for packet speech on the ARPANET, SATNET, and wide-band system will be described.

II. Packet Speech Concepts and Issues

The purpose of this section is to set a general framework for the descriptions of specific experimental packet speech systems to follow in subsequent sections.

A. Generic Packet Speech System Configuration

A generic packet speech system configuration is depicted in Fig. 1. The interface between the user and the network is provided by a functional unit referred to as a packet voice terminal (PVT) [22]. The PVT may, but need not, be implemented in a single physical unit dedicated to a single voice user. Functionally, the user interfaces with the PVT much as with an ordinary telephone set, and the PVT interfaces with the packet network. In addition to being able to talk and listen, the user is provided with a full range of control and signaling capabilities including dialing and ringing. Both control signals and voice are transmitted from PVT to PVT over the network in digitized packet form. The resources of the integrated voice/data packet network are shared statistically with data traffic among host computers and data terminals as well as with other voice users. The packet network may be of the original store-and-forward type as exemplified by the ARPANET; may utilize packet radio, cable, or satellite techniques; or may be composed of an internetwork combination of these various types of packet nets, connected by gateways.

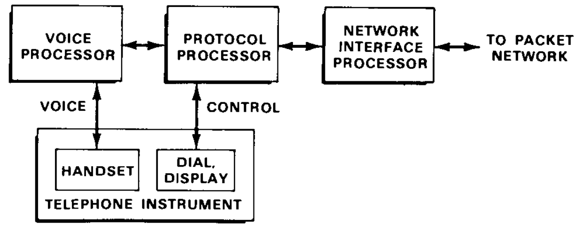

Fig. 2. Functional block diagram of packet voice terminal.

B. Generic Packet Voice Terminal Configuration

A functional block diagram of a packet voice terminal is shown in Fig. 2 which shows three major functional modules each associated with a processor. It is not necessary to use separate processors to achieve the functional modularity, but we have done so in the microprocessor PVT implementation [22] discussed later and find it convenient to use the same terminology here. The voice processor converts between analog and digital speech at digitization rates typically varying from 2 kbits/s to 64 kbits/s, and marks each *parcel* (typically 20–50 ms of speech) as containing either active speech or silence.

The protocol processor is the primary controlling module of the PVT. The protocol processor includes an interface with the user dial/display and must generate and interpret the packets necessary for establishing the call. The protocol processor provides the basic interface between the synchronous voice coding/decoding process, and the asynchronous packet network. The buffering and reconstitution algorithms to produce steady speech to the listener are implemented in the protocol processor.

The network interface processor provides the network-dependent packet transport mechanism. Ideally, all network-dependent hardware and software would be contained in this module. In practice, we have found it difficult to maintain this pure modularity because of a need to incorporate network-dependent elements into the packetization and reconstitution processes in the protocol processor.

The telephone instrument provides the simplest user interface to the PVT. The flexibility of the packet system allows the possibility of a wide range of user functions and displays, which can exceed the signaling capability of the telephone instrument. In some experiments, computer terminals have been used to augment the user interface.

An important development in the work we will describe on packet voice is the evolution of the PVT from implementation on large general-purpose computers to compact microprocessor-based systems. In our view, this development is essential in making packet voice practical and affordable. We have generally focused on a distributed approach where each separate PVT performs complete voice processing and protocol functions for one user. A more centralized approach is also possible, where a single facility would simultaneously perform the functions of a number of PVT's for multiple users.

C. Digital Speech Processing Functions

The primary voice processing function for packet speech is speech digitization. Two other important voice processing functions are also noted here—speech activity detection and echo control.

1) Speech Encoding Algorithms: Speech is a compressible source [34] that can be coded at rates ranging from 64 kbits/s to below 2.4 kbits/s. Recent packet experiments have made use of the pulse code modulation (PCM) widely used in digital telephony, but all the earlier work described in this paper used encoding techniques [36] such as CVSD (continuously variable slope delta modulation) or LPC (linear predictive coding [37]) to provide data rates low enough for use on the networks that were available for experimentation.

Packet systems offer flexibility for taking advantage of speech encoders at a variety of rates. The PVT may include a variety of (fixed) speech bits rates, which could be selectable at dialup according to network load. More complex coding schemes [42] can be applied which vary transmission rates according to the time-varying compressibility of the speech signal. Or multirate "embedded coding" algorithms [38], [39] can be used to allow rapid adaptation [33] of voice bit rates to network conditions which may vary during a call. Selection of a speech coding algorithm [35], [36] for a given application depends on many factors including network bit rate constraints, speech quality needs, noise or distortions on the input speech, and terminal cost and complexity constraints.

2) Speech Activity Detection: A key advantage of packet speech is the ability to save bandwidth by transmitting packets only during talkspurts. Therefore, accurate discrimination between speech and silence, or speech activity detection (SAD), is an essential voice processing function [43]–[45]. The SAD algorithm must minimize the average percentage activity, but also meet tight constraints on the fraction of lost speech. SAD, in a laboratory or quiet input speech environment, is relatively straightforward. But when the speaker is in a noisy environment, or when the speech originated in the switched telephone network (STN), the design of effective SAD algorithms is more difficult.

In our system model, SAD is performed in the voice processor, which marks parcels delivered to the protocol processor as silence or speech. The protocol processor would normally packetize and transmit only the speech parcels except that it may transmit additional parcels at the beginning and end of a talkspurt to improve speech quality. Such a "hangover" at the end of a talkspurt is commonly used to include weak final consonants in a talkspurt and to bridge across short gaps.. An "anticipatory" parcel at the start of a talkspurt can give a smoother startup and is easy to provide in a packet system since the required buffer space is already present for use in the packetization process.

3) Echo Control: Echo control is not needed in a pure packet speech system in spite of the delays that may be present since the system is fully digital and provides isolation between the two directions of voice transmission for the entire path between sending and receiving handsets. However, echo control becomes an issue if we wish to interconnect a packet network and the common STN. Techniques for controlling echos [46], [47] include 1) echo suppression, generally aimed at passing speech in only one direction at a time; and 2) echo cancellation, which attempts to adaptively cancel echos and maintain full duplex speech. Echo cancellation is generally the preferred, but more costly, technique. Echo canceller chips which reduce the cost are becoming available. If the generic PVT were to be used to interface with the STN, it could be equipped with an echo canceller as part of its voice processor, to cope with echoes caused by the two-wire local loop in the STN. Both echo suppression [57] and cancellation [54] have been used in STN interface experiments on the wide-band network.

D. Packet Speech Protocol Functions

The development of the ARPANET as a packet communication resource was quickly, and by necessity, followed by the development of a set of protocols (i.e., rules for conducting interactions between two or more parties) to organize and facilitate use of this resource for a variety of applications. A network control protocol (NCP) was developed to allow controlled packet communication among processes running in dissimilar host computers [9]. Higher level protocols were developed to serve specific user needs. These included TELNET for terminal access to remote computers and file transfer protocol (FTP) for transmission of large files. Both TELNET and FTP obtained access to the network through NCP. This technique of *protocol layering* to partition and organize the task of providing various levels of communication services has been a fundamental aspect of the development of packet communication systems [12].

The original ARPANET protocols were designed to provide very reliable end-to-end packet delivery either at high throughput (e.g., FTP) or low delay (e.g., TELNET). Both NCP and the basic node-to-node protocols imposed end-to-end flow restrictions which included retransmissions when necessary to reliably deliver all the packets and worked against the simultaneous achievement of high throughput and low delay. But for real-time voice communication, both high throughput and low delay are needed. Some reliability may be sacrificed, as a small percentage of lost packets is tolerable. Therefore, new protocol developments were needed for packet voice.

The initial work on packet voice protocols focused around the development of a high-level protocol known as the network voice protocol (NVP). Dr. D. Cohen of the Information Sciences Institute (ISI) was the chief architect of NVP [16], [17]. Functions of NVP include

1) call initiation and termination, including negotiation of voice encoder compatibility and handling of ringing and busy conditions;

2) packetization of voice for transmission, with the time stamps and sequence numbers needed for speech reconstitution at the receiver;

3) speech playout with buffering to smooth variable packet delays.

NVP is designed to pass its packets to a lower level protocol for transport across the network to meet real-time speech requirements. In order to avoid NCP's flow restrictions, NVP bypassed NCP for packet transport. In addition, modifications were made to the basic ARPANET transport protocols to provide an "uncontrolled" packet service which reduced packet flow restrictions between IMP's (see Section IV-B). The original NVP used the basic ARPANET (host–IMP and IMP–IMP) protocols directly to deliver its packets, and was independent of and generally incompatible with other protocols (e.g., NCP) in use at the time.

Since the original NVP made use of the ARPANET directly, extension to other networks (e.g., the Atlantic SATNET) required creation of a new protocol for each new network. This motivated the development of a second generation of voice protocols with a more general internetwork-oriented approach and with network-dependent aspects limited to the lowest level. Protocol functions were separated into two levels. The "higher" functions of call setup, packetization, and reconstitution, as well as dynamic conference control features, were incorporated into a second-generation version of NVP. The lower level protocol, which has come to be named "ST," provides an efficient internet transport mechanism for both point-to-point conversations and conferences. The name ST is derived from the work "stream" which refers to the type of traffic load that voice customers offer to a packet network. ST operates at the same level in the protocol hierarchy as IP, the DoD standard internet protocol [11] for datagram traffic. ST is designed to be compatible with IP. NVP may call on IP for delivery of control packets, and on ST for delivery of voice packets.

ST differs from IP in being a virtual circuit rather than a datagram protocol. Transmission of ST packets must be preceded by a connection setup process arranged by an exchange of control messages. During the connection setup, an internet route is established, and gateways along the path build tables pertaining to the connection. The preplanning involved in the connection setup and the existence of these connection-oriented tables allows ST to offer special services and efficiencies.

Fig. 3 illustrates how the current internet packet voice protocols relate to each other and to corresponding data

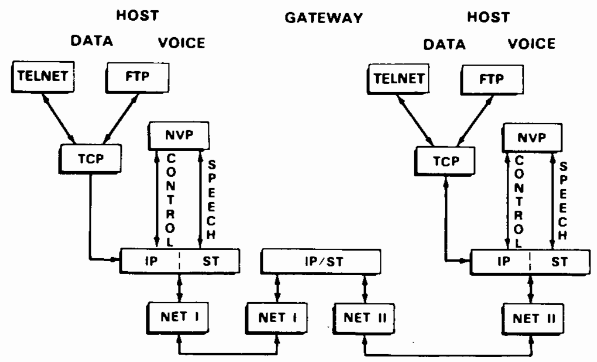

Fig. 3. Protocol hierarchy for internet packet voice and data communication.

handling protocols. Net I and net II designate individual packet networks, and might represent ARPANET, SATNET, or local area cable or radio nets. The situation depicted shows the protocol layers to be traversed in order for voice and data users on net I to communicate (through a gateway) with similar users on net II. The internet data file transfer protocol and the terminal-oriented protocol TELNET utilize a DoD standard transmission control protocol (TCP) for reliable packet delivery. TCP calls, in turn, on IP for packet transport. This is a departure from the original situation in the ARPANET, where FTP utilized NCP, which interfaced directly to the network. Similarly, NVP utilizes both IP and ST for packet transport; IP is used primarily in call setup situations, and ST is used for speech transport.

E. Speech Packetization and Reconstitution

Packet communication necessarily involves both fixed components of delay due to transmission and propagation, and statistically varying components such as queueing delays in network nodes or in gateways. Additional varying delay components are caused by packet retransmissions to compensate for errors in delivery and by the possibility that all packets between a particular source and destination may not follow the same route. In addition to delay effects, some packets may be lost between source and destination. In this regard, a delay versus reliability tradeoff is possible where (for example) delays due to retransmissions can be reduced at a cost of an increase in percentage of lost packets.

The purpose of speech packetization and reconstitution algorithms [31] is to provide speech with 1) minimum overall end-to-end delay and 2) any anomalies caused by lost or late packets basically imperceptible to the listener. Ideally, the overall packet network would provide high enough link bandwidths and sufficient nodal processing power to keep delay and delay dispersion within tightly controlled limits. In such a case, very simple packetization and reconstitution algorithms in the PVT may suffice. However, in some situations where packet speech is required, it may not be possible to control network design. In particular, when there is a need to transmit speech over an existing packet data network, it may be necessary to use more elaborate algorithms.

1) Choice of Packet Size: Resolving the issue of packet size forces us to make some difficult compromises. In order to minimize both the packetization delay at the transmitter and the perceptual effect of lost packet anomalies at the receiver packets should be as short as possible. Experience with lost packet anomalies indicates that individual packets should ideally contain no more than about 50 ms of speech [31]; ideally, we would like packets to be even shorter to minimize packetization delay. On the other hand, in order to maintain high channel utilization, we would like to keep the number of speech bits per packet as high as possible relative to the overhead which must accompany each packet. This tradeoff is particularly difficult for narrow-band speech. For example, 50 ms of 2400 bits/s speech is represented by only 120 bits, which is less than the header size of many existing packet networks. For higher speech bit rates, relative packet overhead is less of a problem. An obvious conclusion is that future packet voice networks should be designed with minimum required header lengths.

The choice of packet size is also influenced by limitations on network throughput in packets/s. For the same user data rate, processing loads on network nodes will generally increase as packet size is decreased. This can force use of longer packets. For example, our typical range of packet sizes for real-time speech transmission across the ARPANET was 100–200 ms, corresponding to 5–10 packets/s because the network could not consistently sustain a higher rate. In some cases it may be desirable to adapt packet size to time-varying network conditions. In speech experiments conducted by SRI on packet radio nets (PRNET's) [20], [21] the radio provides channel availability information to the voice terminal which buffers speech and sends variable size packets depending on the intervals between opportunities for access to the networks.

2) Time Stamps and Sequence Numbers: To assist in the reconstitution process, it is desirable to include a time stamp and a sequence number with each transmitted packet. The time stamp allows the receiver to reconstitute speech with accurate silence gap durations in spite of varying delays between talkspurts. Incorrect gap durations can cause significant perceptual degradation in the output speech, especially for short gaps between syllables, or between words in a phrase. The time stamp also allows reordering of out-of-order packets at the receiver. The time stamp is derived by counting every speech or silence parcel generated by the voice processor. A few bits (we use 12) will suffice to cover a range of relative timing about twice the packet transit time dispersion range of the network.

The sequence number allows the receiver to detect lost packets whereas with a time stamp alone it would not be possible to distinguish silence gaps from packet loss. The detection of lost packets can be used by the receiving PVT to inform the listener (by playing out a distinct audible signal) that some speech has been lost. This can be particularly important if packets contain enough speech to include linguistically significant utterances (such as the word "not"). Detection of lost packets can also be used to allow the terminals to adapt bit rate and/or packet rate to network conditions.

If the network provides service with very short delays and very little delay dispersion, then satisfactory speech can be produced without either time stamps or sequence numbers. However, our experience, both with packet speech experiments and simulations, indicates that both time stamps and sequence numbers should be included.

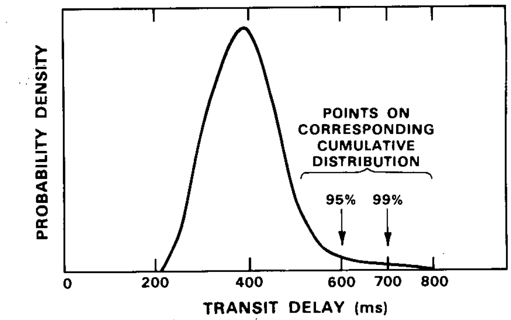

Fig. 4. Illustrative probability density function of transit delays in a packet network.

3) Reconstitution of Speech from Received Packets: The reconstitution algorithm has two major tasks, 1) it must buffer incoming packets and decide exactly when to play them out, and 2) it must decide what to play out when it has finished playing out a packet and the next packet is not available.

Fig. 4 shows an illustrative probability density function for transit delay in a packet network. The delay ranges shown are typical of some of our measurements on 10 hop paths through the ARPANET, but the points to be made are more general. In the case illustrated, 99 percent of the packets experience delays between 200 and 700 ms. Hence, a reconstitution delay (inserted at the receiver) of 500 ms would be sufficient to cover this spread. A 400 ms reconstitution delay would assure playout of 95 percent of the packets. Since some packets may be lost in the net, there is no value of reconstitution delay that can guarantee playout of all packets. Even if all packets did arrive, it would be undesirable to unduly increase delay to account for a few very late arrivals. The network's delay characteristics are generally not known in detail *a priori* and may vary with time. The degree of complexity to be built in to the reconstitution algorithm should be chosen based on the knowledge we do have of the network delays. A fixed reconstitution delay would suffice if network delays and delay dispersion are short. If delays are expected to be large or dispersions vary greatly with the network load, it would be desirable to use an adaptive algorithm (see [31] for an example of such an algorithm) to adjust the reconstitution delay to effect a compromise between packet loss and overall delay.

The other major reconstitution algorithm task is to decide what to play out when it has finished playing out a packet and the next packet is not available. This can result from a late or lost packet or it may simply indicate a pause in the talker's speech. Typically, the reconstitution algorithm has no way to distinguish these cases and should take the same action in either case. A number of fill-in strategies have been tried, including 1) filling with silence, 2) filling by repeating the last segment of speech data, and 3) filling with repeated frames of speech data which are made voice-less and have energy values which decay with time. The third strategy has generally been found to be the most effective, particularly for framed vocoders such as LPC. However, the best choice of fill-in strategy varies with encoder type, packetization size, and statistics of gaps introduced by the network.

F. Conferencing Techniques

Digital voice conferencing imposes a number of requirements in addition to those required for point-to-point speech. There is a need to set up and control multiple connections and to deliver each talker's speech to multiple destinations. If narrow-band speech vocoding is used, a talker selection technique is generally required. Such vocoders cannot successfully handle more than one voice and the alternative of providing several vocoder synthesizers at each site is both cumbersome and expensive.

Packet techniques offer advantages for digital voice conferencing in a number of areas [28]. Since packets need be sent only when speech is present, they can make very efficient use of network resources in conferences where typically only one participant is speaking at any given time. Because connections to packet networks are multiplexed, it is simple for speech terminals and conference controllers to exchange control information at the same time that speech is being transmitted. This out-of-band signaling capability helps in achieving effective conference control, including the control algorithm which selects a talker to "have the floor" at a given time. The use of packets simplifies the implementation of distributed conference control, an important feature for military applications where its use can enhance survivability.

In order to explore the features and problems of packet voice conferencing in some detail, experimental implementations described in sections to follow have been carried out on ARPANET, SATNET, and the WB SATNET.

G. Statistical Multiplexing of Packet Voice and Data

An important goal for packet voice systems is to achieve efficient statistical multiplexing of multiple voice users, and of voice users with data traffic, on common transmission resources. Much analysis and simulation work has been reported showing potentials and limitations of voice/data multiplexing for various system configurations. One of the goals of packet speech systems experiments is to validate these results or identify practical limitations not shown in the analyses.

Some selected observations related to statistical multiplexing in packet voice systems are noted below. These observations and related analyses or simulations are described in [48]. Similar results have been obtained by a number of other researchers [51].

First, packet speech multiplexing allows a straightforward utilization of the tradeoff between delay and channel utilization (or equivalently between delay and "TASI advantage") [49], [50]. The number of users multiplexed onto a link can be increased at a cost in variable buffering delay at the multiplexer. The relative efficiency improvement offered by buffering is greatest where a small number of users are multiplexed, indicating potential for efficiency in a distributed net where local concentrations may be smaller than required for efficient circuit-switched TASI.

A second observation, based on simulations (as cited in [48]), is that interactive data traffic (characterized by Poisson packet arrival processes) can make efficient use of silence intervals in voice calls. However, the utilization by data traffic of varying capacity due to voice call initiation and termination is not nearly as effective due to the much slower variation in channel capacity used by voice.

A third observation is that local area carrier-sense multiple-access (CSMA) cable networks can be used effectively for voice [23]. The bandwidth utilization of such a CSMA network can be equal to or better than the efficiency obtained using fixed time division multiple access (TDMA). CSMA cable networks have been effectively employed for packet voice and are an important part of the experimental wide-band system.

Finally, variable-rate voice flow control techniques [33] using embedded coding can be employed effectively in situations where we are attempting to maintain link loads close to capacity, and temporary overloads are inevitable. Embedded coding allows immediate response by network nodes to such overloads (by discarding packets), with minimal impact on speech users, since communication can be maintained with a temporary degradation in speech fidelity.

III. Summary of Packet Speech Experiments

A summary of key characteristics of the packet speech experiments conducted under DARPA sponsorship is shown in Table I. More detail on each set of experiments will be presented later; for definition of the abbreviations and acronyms used in Table I, see the Appendix. The first network to be used was the ARPANET which consists of intelligent store-and-forward nodes called interface message processors (IMP's) connected primarily by 50 kbits/s point-to-point leased lines. Later, broadcast nets using satellite, radio, and cable were utilized. Initial internetting experiments were conducted using ARPANET and SATNET. The wide-band system is specifically configured as an internetwork where voice users reside on local nets and access the WB SATNET through gateways. Interoperation with circuit-switched telephone systems has also been introduced in the wide-band system. Such interoperation would be essential in introducing packet speech into an environment dominated by circuit-switched voice users.

The link (point-to-point) or channel (broadcast) bit rates quantitatively indicate the limited capacity available for voice in the earlier experiments as well as the greater capacity of the wide-band system. Because of limited network bit rates, most of the experiments on ARPANET and SATNET used LPC vocoding. A few CVSD experiments (primarily at 9.6 kbits/s) were conducted on ARPANET. Voice bit rates used in the wide-band system have ranged

TABLE I
SUMMARY OF PACKET SPEECH EXPERIMENTS

Networks	Network Types	Link or Channel Bit Rates (Kbps)	Voice Algorithms and Bit Rates (Kbps)	Time Period	Sites	Voice Processors	Protocol Processors
ARPANET	Point-to-Point (PTP) Store and Forward (SF)	50	LPC,LPC (VFR): 2-5 CVSD: 9.6-16	1974-79	CHI, ISI, LL, SRI	AP120 AP120B FDP,LDVT SPS-41	MP32 PDP-11/45 TX-2,PDP-11/45 PDP-11/40
SATNET	Broadcast (B'cast) Satellite	64	LPC: 2.4	1977-79	BBN, NDRE, UCL	LPCM	PDP-11/40
ARPANET + SATNET	PTP/SF + B'cast Sat	50 + 64	LPC: 2.4 LPC: 2.4	1978-79	ISI,LL + NDRE,UCL	AP120B,LDVT LPCM	PDP-11/45 PDP-11/40
PRNET	B'cast Radio	100-400	LPC: 2.4, CVSD: 16	1978-83	SRI	LPCAP, CHI-V	LSI-11 PDP-11/23
WB SYSTEM							
WB SATNET +	B'cast Sat +	772-3088	LPC: 2.4,		LL,	CLPC	8085
LEXNET +	B'cast Cable +	1000	CVSD: 16,		ISI,	AP120B	PDP-11/45
PRNET +	B'cast Radio +	100-400	ECVSD: 16-64,	1980-83	SRI,	CHI-V	PDP-11/23
TELEPHONE NETS	Circuit-Switched	---	PCM: 64		DCEC		

TABLE II
PACKET CONFERENCING EXPERIMENTS

NETWORKS	CONTROL TECHNIQUES		PACKET ADDRESSING	DEMONSTRATED	SITES
ARPANET	CENT	PB	PTP	1976	CHI, ISI SRI, LL
SATNET	DIST	PB	B'CAST	1978	NDRE, UCL, BBN
ARPANET + SATNET	CENT + DIST	PB	PTP + B'CAST	1979	ISI, LL + NDRE, UCL
SATNET	DIST	VOICE	B'CAST	1979	NDRE, UCL, BBN
WB SYSTEM	DIST	VOICE	B'CAST	1982	ISI, SRI, LL, DCEC

CENT = CENTRALIZED
DIST = DISTRIBUTED
PB = PUSH BUTTON
PTP = POINT-TO-POINT
B'CAST = BROADCAST

from 2.4 to 64 kbits/s. Accommodation of 64 kbits/s PCM is important in allowing convenient interoperation with digital circuit-switched systems which use PCM as a standard.

As indicated, a large variety of narrow-band voice processors and protocol processors have been used in the packet speech experiments. Voice processors range from special laboratory-built programmable signal processors (e.g., FDP, AP120, LDVT), to very compact LPC units (CLPC). Protocol processors include general purpose network host computers (e.g., PDP-11/45) and small microprocessor-based units (e.g., 8085). The trend through the course of the program has continually moved toward smaller size, weight, and power.

The large number of site organizations involved, as well as the associated time periods, are indicated in Table I.

Conferencing has been of major importance in the packet speech experiments, and Table II summarizes features of conferencing experiments which have been carried out. Both centralized and distributed control techniques have been used for conference setup and for determination of which speaker has the floor at a given time. In ARPANET and SATNET, a conferee indicated his desire to talk by pushing a button, and indicator lights were used to inform the conferee that he had the floor. In later systems, a conferee could try to gain the floor by beginning to talk. A voice-controlled floor controller provided arbitration among multiple talkers. The voice-control strategy gener-

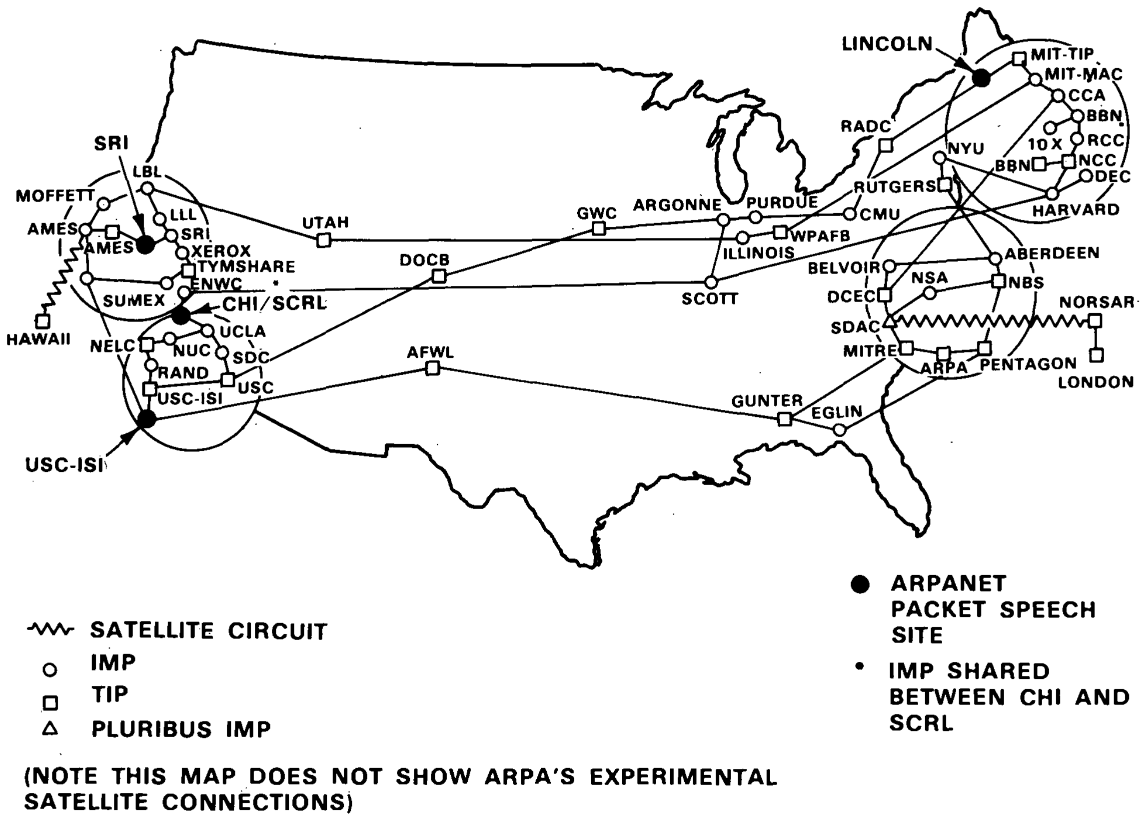

Fig. 5. Geographic map of the ARPANET, as of June 1975, showing locations of ARPANET packet speech sites.

ally gave more satisfactory performance from a human factors point of view [64]. The packet addressing mode is also important in conferencing. A broadcast mode avoids replication of voice packets or of conference control packets for multiple receivers.

IV. Packet Speech on the ARPA Network

A. ARPANET Characteristics

The ARPANET is a large store-and-forward packet-switching network [1] which interconnects computer facilities at a variety of locations. The network has been growing and evolving constantly since its initial four-node operation late in 1969. A June 1975 network map, representative of the topology in effect when most of the packet speech experiments were performed, is shown in Fig. 5. The network sites involved in the speech experiments were CHI, ISI, and SRI on the West Coast, and LL on the East Coast. The intersite distance among these locations (in number of hops on the shortest path) generally varied from 5 to 10.

Each ARPANET node generally consists of a communications processor called an interface message processor (IMP) developed by BBN. The IMP's are connected by 50 kbit/s lines according to the indicated topology. Host computers connected to the IMP's at each site deliver "messages" to the network with headers indicating the destination address. Depending on the number of bits in the message, it will be transmitted across the network by the IMP's as one or more ARPANET packets. The IMP's route each packet independently to the destination. As packets travel through the net on the lines between IMP's, they carry a packet header of approximately 160 bits. The maximum amount of user data that can be carried with each such packet is approximately 1000 bits.

B. Speech Transport in the ARPANET

The ARPANET characteristics lead to upper bounds on speech throughput due to the 50 kbit/s links and the transmission overhead, and lower bounds on delay due to the multiple hops generally required between source and destination. In addition, the original protocols developed for the ARPANET included reliability and flow control features which were designed appropriately for data communication, but which caused undesirable and unnecessary limitations on the throughput and delay for real-time speech. These limitations were present both in the packet delivery service provided by the IMP subnet between source and destination host, and in the original host/host or network control protocol (NCP) used in the ARPANET. Because of these limitations a new host/host protocol (NVP) was developed for speech and a new type of "uncontrolled" packet delivery service (suggested by Dr. R. E. Kahn) was introduced into the ARPANET.

The original NCP protocol implementations [9] generally allowed only one "message" at a time to be in flight between a pair of processes in a source and destination host. The next message would not be sent until an acknowledgment, known as a request-for-next message (RFNM), was received from the destination. One motivation for the message-at-a-time limitation was to prevent a single user process in a multiuser host from dominating the host/IMP line. This "fairness" criterion was in conflict with the need to provide priority service to speech users. Messages could include up to 8063 bits of user data. Any message larger than the maximum packet size of 1008 bits would be broken up by the source IMP into a multipacket message to be transmitted across the net and reassembled by the destination IMP. High throughput could be attained by sending large multipacket messages. This is reasonable transport service for file transfers but sending multipacket messages for speech results in an undesirably large packetization delay. On the other hand, single-packet messages allow lower delay but result in severe throughput penalties, particularly for a path containing many hops. For example, a typical minimum round-trip time to send a 1000 bit single-packet message across a 10 hop ARPANET path and to receive a RFNM is about 0.3 s. The resulting peak throughput for the "message-at-a-time" protocol is 1000/0.3 = 3333 bits/s with the average being significantly lower. Because of these restrictions NVP bypassed the NCP protocol modules which were available at the time when the network speech experiments were initiated and instead interfaced directly to the IMP subnet through the host/IMP protocol.

But the IMP subnet itself imposed important limitations on speech traffic. First, a restricted number of messages was allowed to be in flight between source and destination IMP's without a RFNM being received. When speech experiments started this number was 4; it was increased to 8 late in 1974. This restriction was imposed by IMP buffer space. More fundamentally, the IMP subnet provided reliable in-order end-to-end delivery of messages. If any message was lost and had to be retransmitted, all subsequent messages would be delayed to wait for the successful retransmission. This characteristic was reasonable for data terminal or file transfer traffic, but for speech it caused an occasional late packet to result in lengthy glitches. Fortunately, the rarity of packet errors in ARPANET did allow some successful speech communication despite this error control and sequencing.

For the above reasons, the new "type 3" packet delivery service was incorporated into the ARPANET by BBN on an experimental basis late in 1974. This new service allowed single-packet messages to be transmitted between selected hosts without end-to-end error control, without sequencing, and without a restriction on the number of packets in flight. This mechanism was used for most of the ARPANET packet speech experiments. Most of those experiments were conducted in conditions of light network loading. Use of type 3 packets in heavy load conditions was restricted to avoid the possibility of ARPANET congestion affecting all users.

Fig. 6 shows a comparison of cumulative round-trip delay distributions for type 0 (ordinary service with controls as described above) and type 3 messages between Lincoln and ISI (10 hops at the time the data were taken), taken in June 1975. Each message consisted of a single packet with approximately 1000 data bits. Packet rate was 8.6/s for net user bit rate of 8.6 kbits/s. At a 1 percent lost packet rate, type 3 is seen to provide about a 0.4 s advantage in overall delay. For higher rates type 0 became unusable whereas it was possible at the time to support 16 kbit/s CVSD with type 3 packets (but only during hours when network load was light). For lower rates, such as 2.4 kbits/s, the difference between type 3 and type 0 diminished. Currently, the ARPANET is much more heavily loaded and the results would change accordingly. Additional measurement results on ARPANET speech transmission are reported in [19].

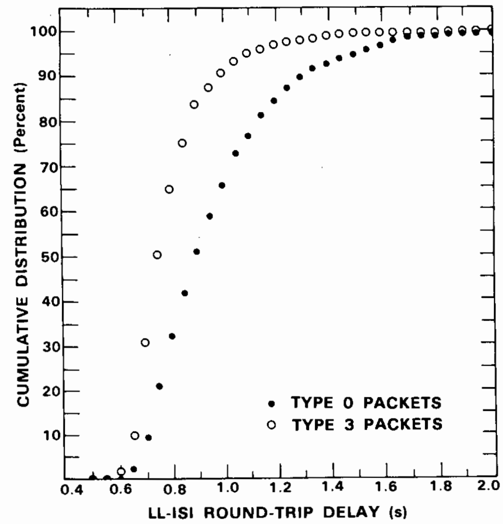

Fig. 6. Comparison of cumulative distribution of round-trip times for type 0 and type 3 ARPANET packets between LL and ISI (10 hops), measured in June 1975. Packet rate was 8.6 packets/s, with 1000 data bits/packet. Minimum around-trip delay on the path was observed to be about 0.6 s.

C. ARPANET Speech System Implementations

ARPANET speech systems were implemented at four sites, as indicated in Fig. 5 and Table I. All sites used different equipment but worked to a common NVP [16] specification. The success in bridging the gap among the systems was an important result in packet voice protocol development, and was achieved through the cooperation of many people in the ARPA packet speech community. The ARPANET speech systems were implemented in minicomputers such as the DEC PDP-11/45 to handle protocol processing with attached programmable signal processors to implement the speech encoding algorithms. Computer terminals were used for controlling call setup, and at some sites high-quality microphones and headphones were used instead of the conventional telephone handset. All sites had measurement software to record system performance. Much of the effort involved in these implementations went into programming the LPC encoding algorithms which were being developed during the same period. Several versions of LPC at data rates from 5.0 kbits/s down to about 2.0 kbits/s were implemented and tested. An ARPANET cofferencing system was implemented with centralized floor control under a CHAIRMAN program running at one site. Conferees had pushbuttons to indicate desire to talk and lights to indicate when they had obtained the floor.

D. Milestone ARPANET Speech Experiments

The earliest packet-speech-related experiments on the ARPANET were conducted by Lincoln Laboratory in 1971 [14] using the TX-2 computer. Speech was not actually transmitted over the ARPANET, but an arrangement was set up whereby two persons could converse while experiencing in real time the effects of packetization and ARPANET delays. Speech was digitized (PCM) and stored. ARPANET delays were introduced by forming messages corresponding to blocks of speech and transmitting to a "fake host" at some IMP in the ARPANET. The fake host would discard the message and return an acknowledgment. Receipt of the acknowledgment was used to indicate that the corresponding block of data could be reconstituted at any time thereafter. Simulated speech bit rates from 2400 to 16 000 bits/s were used. Tests were performed on the effects of vocoder rate, block size, network distance (in hops), and reconstitution strategy. It was concluded that packet speech in a system with characteristics similar to a lightly-loaded ARPANET could be quite satisfactory from a human factors point of view.

The initial milestone in actual packet speech communication across the ARPANET was between ISI and LL, using 9.6 kbits/CVSD, in August 1974. CVSD quality at 9.6 kbits/s is quite poor, but the ARPANET was not capable of supporting 16 kbits/s at that time (Type 3 packets were not yet available.), and narrow-band vocoders were not available for use. In this and all other experiments, the average bit rate was reduced by transmitting packets only during talkspurts. In December 1974, the first LPC speech was communicated at 3.5 kbits/s over the ARPANET between LL and CHI. LPC conferencing at 3.5 kbits/s was first demonstrated in January 1975. Sites involved were CHI, ISI, LL, and SRI; all used different speech processors and host computers (Table I). In April 1978, LPC conferencing was demonstrated using a variable-frame-rate LPC [42] operating in the 2–5 kbits/s range. A 2.4 kbit/s LPC-10 for the ARPANET, first implemented at LL in 1979, was used for ARPANET/SATNET experiments and was later used for LPC experiments in the wide-band system. In addition to the real-time packet speech tests, a variety of experiments [59], [60] were also conducted in person–computer interaction by voice over the ARPANET.

V. Packet Speech on the Atlantic Packet Satellite Network

A. SATNET Characteristics

The Atlantic packet satellite network (SATNET) [4] is a packet-switched network that utilizes a distributed-control demand-assignment multiple-access (DAMA) algorithm called priority-oriented demand assignment (PODA) [2] to share a 64 kbit/s INTELSAT channel among earth stations in the United States and Europe. PODA in SATNET provides several important services for packet voice. First, it offers a type of service called a packet stream which can provide a guaranteed (except for priority preemption) data rate independent of network load. The stream service allows high utilization of the channel and minimizes the effect of network congestion on speech quality. Second, multiaddress packet delivery is provided in SATNET. This reduces the communication costs associated with voice conferencing by avoiding the need to send multiple copies of speech packets. Finally, a datagram service is provided in addition to the stream service. Data service involves the sending of a reservation request message via the satellite. As a result, packets with datagram service experience a cross-net delay at least 250 ms longer than that seen by packets traveling in streams. This type of service was used for control packets to avoid conflicts with the voice stream.

B. SATNET Speech System

Packet speech efforts on SATNET focused on voice conferencing [28] to take advantage of the multiaddress delivery capability. LPC speech at 2.4 kbits/s was used due to the limited bandwidth. The SATNET conferencing programs were designed to use the above features and also to explore the potential for distributed conference floor control. In a satellite net, distributed floor control achieves a delay advantage over centralized control of at least one satellite roundtrip.

In SATNET conferencing, the conference control programs (CCP's) at each site shared a common uplink stream to minimize use of capacity. On the downlink, stream packets were addressed simultaneously to all CCP's including the sender. The CCP's controlled access to this stream on a distributed basis. Communication of control packets was carried out via broadcast datagrams. Datagrams among CCP's were also used to resynchronize the conference when control errors occasionally caused two or more talkers to collide in the shared stream. Such collisions would be detected by the CCP receivers and recovery would be initiated.

Participants in the initial SATNET conferences were provided with a conference-control box equipped with push buttons and lights. A participant desiring to talk would push a want-to-talk (WTT) button which would cause a WTT message to be broadcast to all CCP's. On receiving that message, each CCP would add the participant to a WTT queue. Pushing a DONE-TALKING button would relinquish the floor by sending a control message in the voice stream. All CCP's assumed the head talker in the WTT queue to be the next speaker. Pushing the DONE-TALKING button would also remove a waiting participant from the WTT list.

A later version of SATNET conferencing employed voice control using SAD. A participant was allowed to transmit speech packets when none had been received within the last half second. A preassigned priority was used to resolve

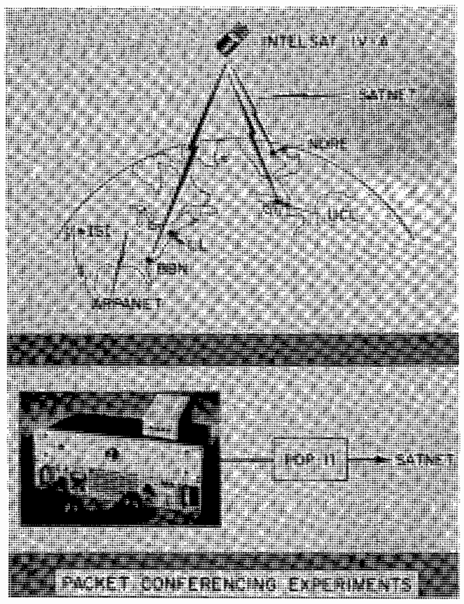

Fig. 7. Configurations of sites and equipment for SATNET and SATNET/ARPANET packet speech experiments.

collisions. Human factors studies [64] have concluded that voice control is preferable since it is easier to learn. Also, the queue associated with the push-button control sometimes leads to a "town meeting" effect where participants join the queue and then rehearse their speech instead of listening.

Hardware and software to support SATNET conferencing were developed by Lincoln Laboratory and installed at NDRE, UCL, and BBN. Hardware included a linear predictive vocoder [40], a PDP-11 interface, and a conference control box, all shown in Fig. 7. Voice protocol and conferencing software were implemented in PDP-11 SATNET host computers residing at the sites. Fig. 7 also shows the locations of SATNET conferencing sites and of the sites involved in SATNET/ARPANET internetwork conferencing.

C. SATNET/ARPANET Internetwork Speech System

To support internet conferencing, software was written for the SATNET host computer at BBN which also served as a gateway to ARPANET. The software made the BBN PDP-11 act as a special conferencing gateway. It functioned both as a participant and as the central controller in an ARPANET conference and as a participant in a simultaneous SATNET conference. Vocoder programs were written for the ARPANET sites to match the hardware vocoders at the SATNET sites. This internet system deomonstrated operation of a combination of centralized control and point-to-point packet delivery in the ARPANET with distributed control and broadcast delivery in SATNET. However, it pointed out the need for a more general approach to internetting since it was necessary to have very specialized software running in the gateway to deal with the different protocols in effect in the two nets. The new voice protocols developed for the wide-band network eliminate much of this specialization.

D. Experimental Results and Milestones

SATNET conferencing among the three sites using push-button control was first demonstrated in May 1978. The later version using voice control became operational in November 1979. Internet conferences were first carried out in September 1979 with SATNET participants at NDRE and UCL and ARPANET participants at LL and ISI. These systems have demonstrated the technical feasibility of packet voice conferencing in existing packet networks. SATNET conferencing, in particular, has demonstrated that the survivability advantages of distributed control can be achieved with little loss in conferencing performance.

VI. Packet Speech on the Experimental Wide-Band System

A. Introduction and System Overview

An experimental wide-band satellite-based packet system [52], [53] has been implemented to develop and demonstrate techniques for integrating packet voice with data in a realistic large scale network. The system is designed around a satellite channel with a capacity of 3.088 Mbits/s, in order to support many simultaneous voice connections. Whereas the ARPANET and SATNET were fundamentally data networks, on which limited speech experiments were performed, the wide-band system was designed specifically to accommodate speech. The wide-band system is configured as an internetwork where voice users reside on local networks and obtain access to the wide-band packet satellite network (WB SATNET) through gateways. This introduces a useful multiplexing hierarchy where traffic from local sources is first multiplexed by local nets and gateways, while the WB SATNET nodes in turn multiplex the satellite channel among aggregated traffic sources from the gateways at all the nodes.

The wide-band packet speech system development and the experimental program are sponsored by DARPA and involve a cooperative effort among a number of organizations as cited below. The Defense Communication Agency (DCA) has sponsored the satellite network development along with DARPA, and is utilizing the WB SATNET for a set of experiments supporting the development of the future defense switched network (DSN) [62], [63]. One of the four original network nodes is located at the Defense Communications Engineering Center (DCEC) in Reston, VA.

B. The Wide-Band Packet Satellite Network

The WB SATNET is a higher performance version of the Atlantic SATNET described in Section V-A. It uses the same DAMA algorithm (PODA) to share a 3.088 Mbit/s channel. The channel on the WESTAR III satellite and the earth stations are leased from Western Union, Inc. WB SATNET differs from the Atlantic SATNET in the use of earth stations with smaller antennas and has link budgets that result in bit error rates at 3.088 Mbits/s that require forward error correction of control packets to maintain

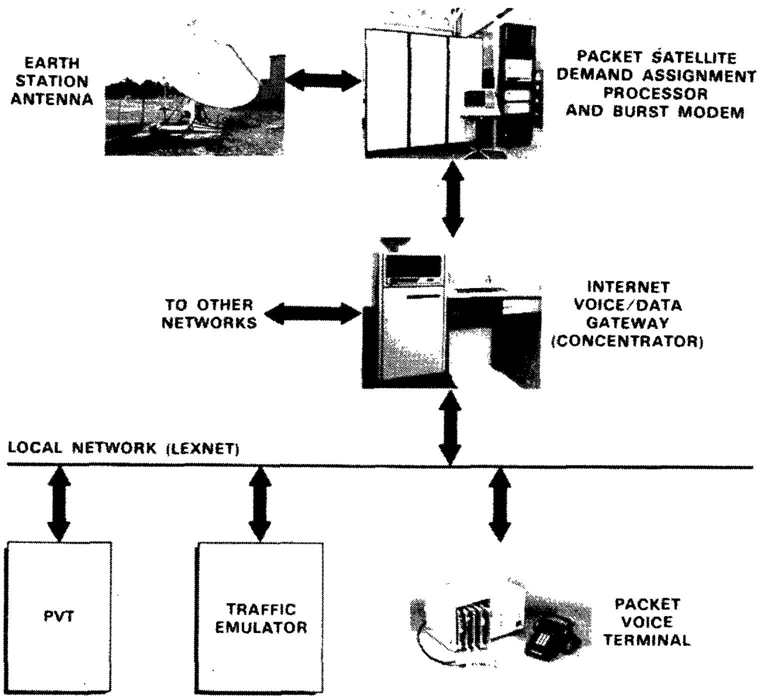

Fig. 8. Equipment configuration for typical wide-band network site.

synchronization of distributed PODA controllers. The earth station interface equipment provides multirate error correction to support this requirement. This error correction can also be applied to user data at the option of the user with consequent reduction in net data rate. The WB SATNET equipment at each site includes three major subsystems, a satellite earth station, a flexible burst modem called an ESI (earth station interface, developed by Linkabit, Inc.), and a packet satellite DAMA processor called a PSAT (pluribus satellite imp, developed by BBN) [55]. These WB SATNET subsystems are illustrated in Fig. 8 which also shows a traffic concentrator (i.e., a gateway) and a local net at the Lincoln site.

Features of the WB SATNET which are of interest for packet speech experiments are: 1) a sufficiently wide-band channel to support multiple voice users, even without narrow-band speech coding; 2) the capability for multiple coding rates to accommodate the different bit error rate requirements of speech and control packets; 3) stream reservations on the channel to provide guaranteed data rate and minimum (i.e., one hop) delay for speech; and 4) broadcast capability for efficient voice conferencing.

C. Wide-Band System Speech Facilities and the ST Protocol

Fig. 9 shows a map of the wide-band internetwork system, focusing on the primary local area facilities at each site. Internet voice/data gateways (G) based on a DEC PDP-11/44 minicomputer have been developed by Lincoln Laboratory and have been used for most of the wide-band system experiments. These gateways (Fig. 10), also referred to as "miniconcentrators," support both the experimental ST protocol and the DoD standard IP protocol. Key speech-related ST functions include obtaining satellite channel stream allocation based on local user bit rate requirements and concentrating speech packets from local terminals into aggregated packets for the WB SATNET. Table III lists major requirements for efficient packet speech transmission along with the approach used in ST to meet these requirements. Satellite channel allocation requests are ideally set on a statistical basis taking account of the fact that voice is transmitted only during talkspurts. The development of ST has been a major facet of the wide-band program. Although ST operates at an internet level in the wide-band system, the approach is valid for an individual network [29]. Gateway ST functions would be performed by network nodes in an individual net.

The PDP-11-based gateways are multiported and can provide simultaneous connections to more than one local net. Measurements have indicated an available throughput of 600–900 packets/s depending on packet lengths. More than one gateway can connect to a PSAT; the LL and ISI sites have both miniconcentrator gateways and a BBN-developed very high throughput multiprocessor concentrator/gateway referred to as the voice funnel [56].

Local broadcast cable networks (referred to as LEXNET's, for Lincoln Experimental Networks) [22]–[24] were developed at LL to efficiently support local packet voice and data traffic. LEXNET's have been installed and operated at all four sites. LEXNET is a 1.0 Mbit/s baseband cable network with distributed control, which uses a carrier-sense multiple-access protocol with collision detection (CSMA/CD) similar to that used in Ethernet. It utilizes a distributed algorithm for randomized retransmission which is specialized for voice traffic and which has been shown by simulation studies to provide high channel utilization for voice. The algorithm estimates competing

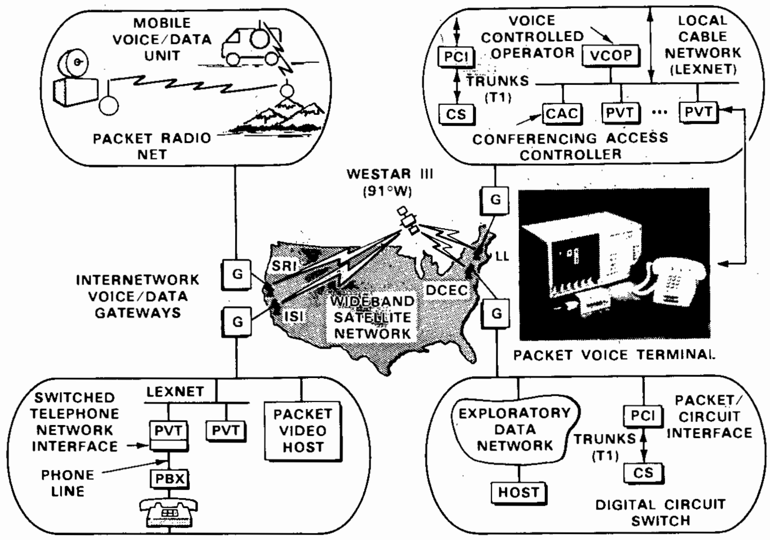

Fig. 9. Wide-band internetwork packet voice/data system, with illustration of primary local area facilities at each site.

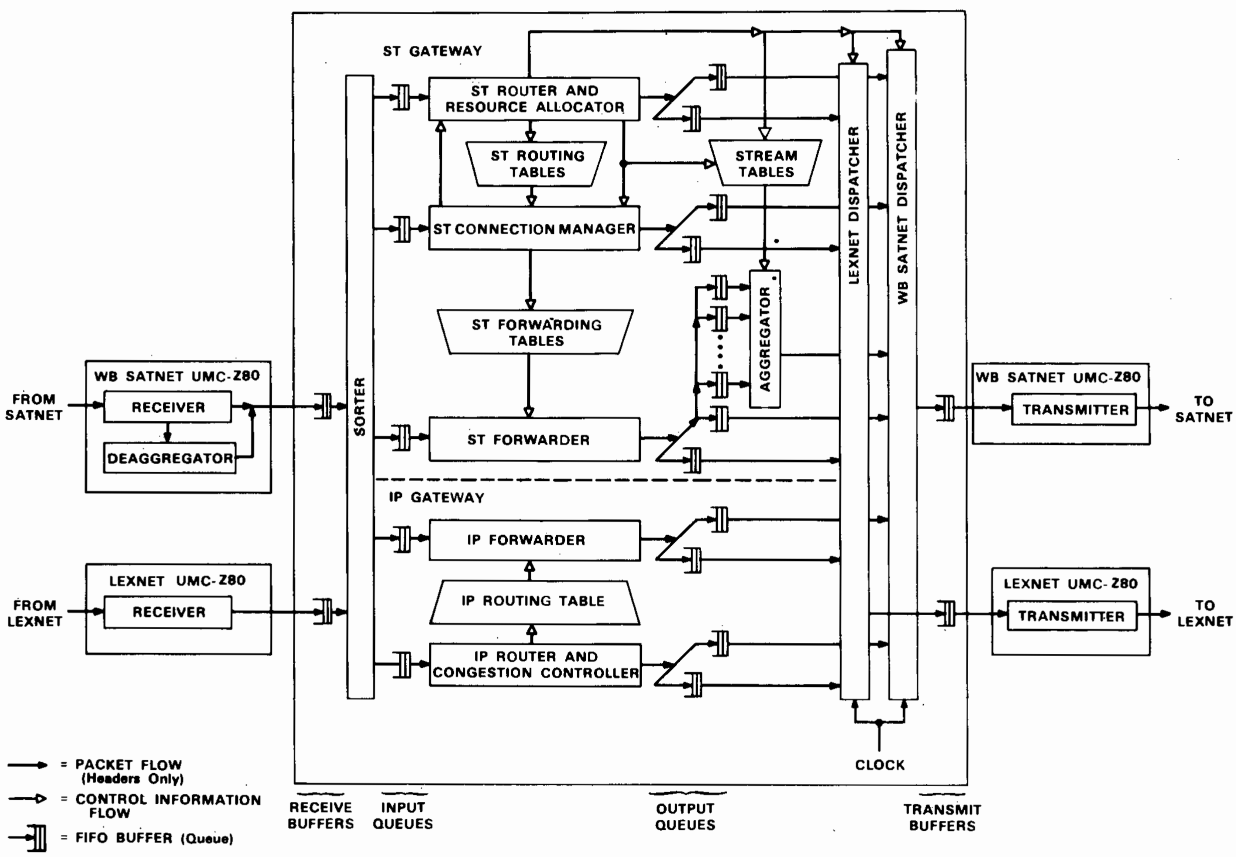

Fig. 10. Block diagram of miniconcentrator gateway. A PDP-11 central processor is used, and network interface processors (UMC-Z80 boards produced by Associated Computer Consultants) are included with special hardware interfaces for each attached network.

network activity and adjusts its retransmission interval based on the fact that voice terminals produce periodic packets during talkspurts. LEXNET's are populated by compact, microprocessor-based packet voice terminals (PVT's) [24] which provide full voice processing and protocol functions (see Fig. 11). The PVT's support 64 kbit/s PCM voice digitization or a choice of lower rate plug-in vocoders. In particular, Lincoln-built single-card 2.4 kbit/s LPC [41] and 16–64 kbit/s embedded CVSD (ECVSD) [39] units are available for experiments.

Conferencing using the second-generation voice protocols requires the services of a central access controller to

TABLE III
THE ST PROTOCOL FOR PACKET SPEECH

Packet Speech Requirements	ST Approach
1) Guaranteed data rate.	1) Know requirements in advance. Request reserved network resources when available (e.g., PODA streams). Assign loads to links statistically in routing virtual circuits.
2) Controlled delay (predictable dispersion).	2) Prevent congestion by controlling access on a call basis.
3) Small quantity of speech per packet.	3) Set up virtual circuit routes so that abbreviated headers can be used. Aggregate small packets for efficiency.
4) Efficiency equal to or better than circuit switching without TASI.	4) Abbreviated headers for packet efficiency. Goal of high link utilization with effective traffic control.
5) Efficient use of broadcast media.	5) Control multiaddress setup for conferencing and replicate packets only when necessary.

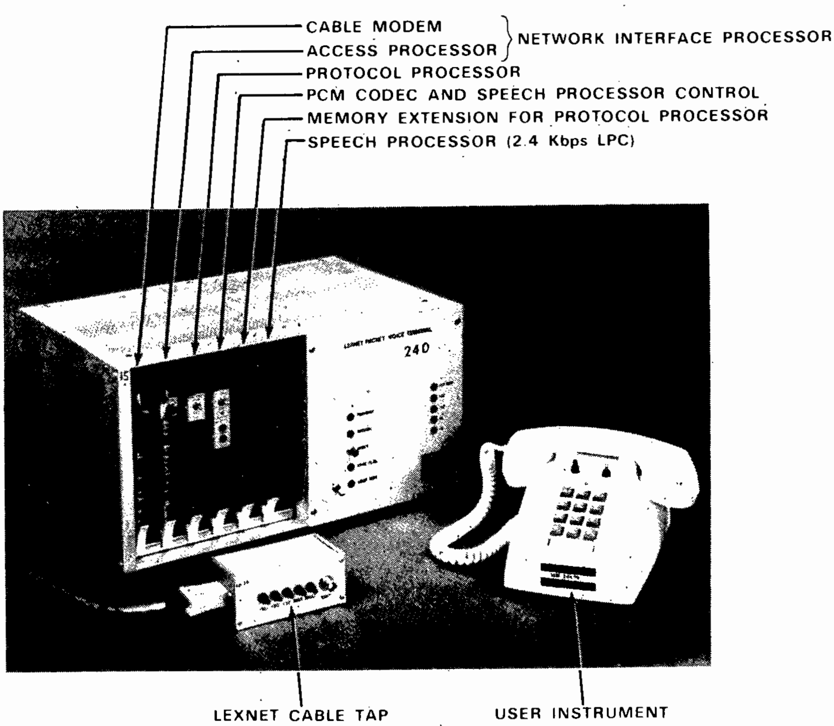

Fig. 11. Lincoln packet voice terminal. The three primary functional units are each controlled by an Intel 8085 microprocessor. The LPC unit utilizes three high performance signal processing microcomputers for analysis, synthesis, and pitch detection. The protocol processor supports NVP and ST and has a general interface to the access processor to allow adaptation to other networks. The user instrument has an 8085 which controls ringing, dial tone, etc. The PVT package is composed of approximately 200 integrated circuits, consumes 40 W, and occupies 0.75 ft^3 of volume.

assure uniqueness of conference connection identifiers throughout the network and to regulate access to particular conferences according to instructions provided by the conference originator. These functions are performed by the conferencing access controller (CAC) that resides on the LL LEXNET (the CAC address is assumed to be known to all PVT's and need not be dialed by users). The CAC is involved only in the process of setting up and taking down conferences and plays no part in the dynamic control of the conference "floor." It is implemented using PVT

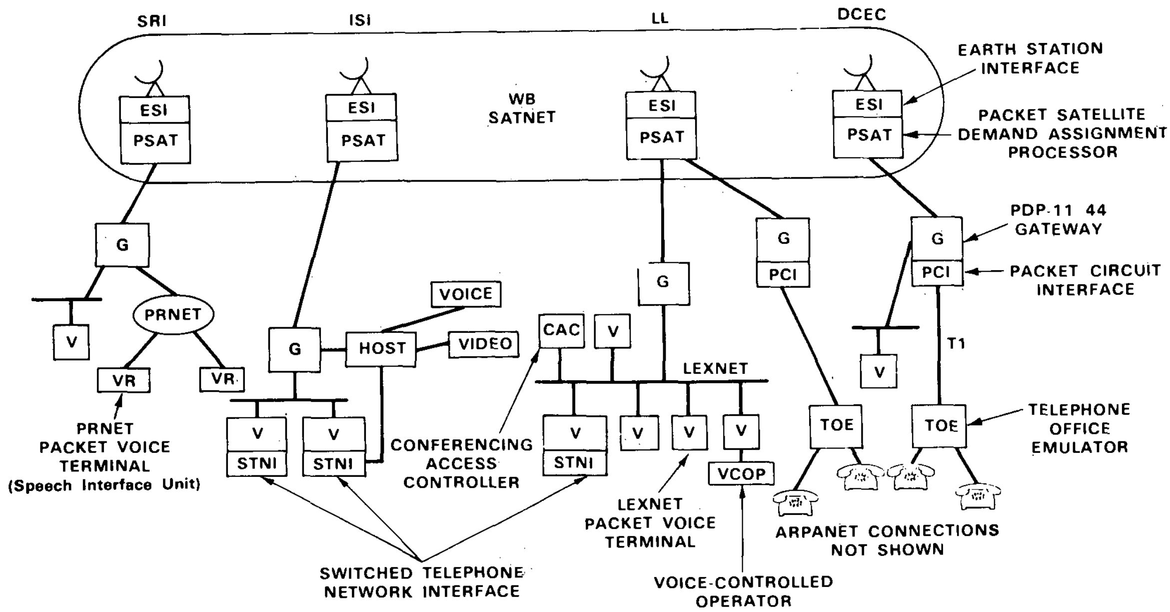

Fig. 12. Wide-band packet speech experiment status—September 1982.

hardware with special CAC software running in the protocol processor. A voice-controlled operator (VCOP), which allows conference setup via dialog with speech recognition and synthesis devices, is also resident on a LEXNET at LL [61].

The packet radio network (PRNET) located in the San Francisco Bay area [7] includes both fixed and mobile units, and both voice and data terminals. PRNET voice terminals [20], [21] include PDP-11/23-based speech interface units (SIU's) which implement voice protocols; speech coding is accomplished via 16 kbit/s CVSD units or CHI-5 2.4 kbit/s vocoders. Packet routing from the mobile PRNET to SRI can switch automatically as required from line-of-sight to double connectivity via hilltop repeaters. The PRNET, primarily designed for data, can support only limited voice traffic. But PRNET voice experiments have led to definition of a new PRNET type of service for better service of real-time voice [21]. In particular, voice service can be improved by allowing voice routes to change more rapidly than routes for data traffic.

Two kinds of interfaces are shown between the packet-switched network and circuit-switched systems. ISI has developed a switched telephone network interface (STNI) [57] which allows connection from individual telephone lines to the wide-band packet system. The STNI takes the form of a card which resides in a LEXNET PVT and allows the user to dial into the wide-band system from any ordinary telephone by first calling the STNI, which provides a second dial tone and accepts dialed digits addressing other PVT's. The STNI card handles translation of dialing and analog voice between the PVT and the public net, provides PCM digitization, and includes echo suppression. STNI's are currently installed at LL as well as at ISI. A packet video facility has also been developed by ISI to support low rate packet video experiments.

The packet/circuit interface (PCI) was developed by Lincoln under DCA sponsorship [54] to allow communication between packet switches and digital circuit switches in the T1 digital carrier format used for multiplexing of interswitch trunks in digital telephony. Telephone office emulators (TOE's) are provided to simulate the traffic from local digital circuit switches. The PCI is primarily being used for experiments in which a DAMA satellite is used as an overlay to a terrestrial circuit-switched net. These experiments are being carried out under DCA sponsorship to develop networking techniques applicable to the planned defense switched network which will utilize a mix of satellite and terrestrial media to provide survivable and economical telecommunications for DoD subscribers. The PCI/TOE facility has also been used to demonstrate interoperability between circuit-switched users (i.e., telephones on a TOE) and packet voice users on LEXNET PVT's. Each PCI provides up to four 64 kbit/s PCM trunks. In translating from T1 to packet format, the PCI must implement a subset of NVP and ST. The PCI thus performs the functions of a multiuser PVT, and in fact, carries out the protocol functions (both call setup and transport) for four simultaneous users. Special four-wire phones are provided at each TOE, but a COMSAT telesystems echo canceller is provided for access from standard two-wire phones. At DCEC, a gateway connection to an exploratory packet data network (EDN) is provided to help support packet data experiments in the wide-band system.

D. Experimental Results and Milestones

A snapshot of the wide-band internetwork packet speech system, as configured in September 1982, is shown in Fig. 12. All the internet packet speech capabilities implied by that figure have been demonstrated [53]. These include: multiple simultaneous PTP calls using PCM, ECVSD, and LPC; PCM and LPC conference calls using distributed floor control; voice internetting among LEXNET's, PRNET, and cirucit-switched systems; and conference setup using VCOP. The new internet ST protocol has been implemented and tested successfully both in gateways and in terminals. Interoperation between miniconcentrator and voice funnel gateways has been demonstrated. Compatible LPC voice processing and NVP/ST protocols (both point-to-point and conferring) have been implemented in LEXNET PVT's and in PRNET SIU's.

The earliest major milestone in the achievement of the packet speech internet testbed capability occurred in November 1981 when two simultaneous PCM conversations were carried over WB SATNET between LL and ISI using PVT's on LEXNET's. One of these calls originated at an ordinary telephone extension at ISI and entered the wide-band system through an STNI. During 1982, the other capabilities were demonstrated: circuit-to-packet interconnection via PCI's in March; communication with a mobile PR terminal and multisite conferencing in June, and voice-controlled conference setup in October.

Current efforts are focused on performance measurements on the wide-band system, building on the basic demonstrated capability for internetting multiple voice users. A combination of real and emulated voice and data traffic is being applied to assess performance breakpoints in local nets, gateways, and the WB SATNET itself.

VII. Discussion and Conclusions

The successful system implementations and experiments described here strongly support the conclusion that packet communication is a practical technique for real-time speech communication. In cases where a user has already invested in a packet data communication network, adding a speech service to this network may well be a more economically attractive alternative than providing a separate speech service.

The great deal of interest in packet speech being shown by telecommunications companies, as evidenced by a number of current publications, including those in this current Special Issue of the IEEE Journal on Selected Areas in Communications, attests to the potential long-term advantages of packet techniques for integrated voice and data communication.

The work described here has provided a practical demonstration of the feasibility of packet speech in a large variety of packet network and internetwork environments. These system implementations have provided stimulus for the definition of packet speech requirements and for the successful development of speech processing techniques, voice protocols, packetization and reconstitution strategies, digital voice conferencing, and voice/data multiplexing. In addition, some of the advanced services possible through integration of voice and computer communication in the same network have been demonstrated, including voice interaction between computers and people in the network environment.

The vast investment in circuit-switched systems currently in existence makes it unlikely that packet techniques will soon become the dominant method for speech communication. However, as illustrated by the circuit/packet interoperability experiments described here, a useful coexistence of circuit-switched and packet-switched speech systems can be achieved. Meanwhile, the use of packet speech can be expected to grow over the next few decades.

Appendix
Acronyms and Abbreviations

AP120	—an early array processor developed by CHI, used for ARPANET speech
AP120B	—commercially-available array processor developed by Floating-Point-Systems, Inc.
BBN	—Bolt, Beranek and Newman, Inc., Cambridge, MA
CAC	—conference access controller
CCP	—conference control program; used for distributed control of SATNET packet speech conferences
CHI	—Culler-Harrison, Inc., Goleta, CA; now known as CHI Systems, Inc.
CHI-V	—array processor developed by CHI
CLPC	—compact LPC; single-card unit developed by Lincoln Laboratory
DEC	—Digital Equipment Corporation
ESI	—earth station interface; developed by Linkabit, Inc.
Ethernet	—CSMA/CD packet data cable network developed by Xerox
FDP	—fast digital processor; digital signal processing computer developed by Lincoln Laboratory
IMP	—interface message processor; the nodal processor in the ARPANET, developed by BBN
INTEL 8085	—microprocessor developed by INTEL Corporation
ISI	—Information Sciences Institute, Marina Del Rey, CA
LDVT	—Lincoln digital voice terminal; a programmable signal processing computer
LEXNET	—Lincoln experimental packet voice network
LL	—Lincoln Laboratory, Lexington, MA
LPC-10	—tenth-order linear predictive coding
LPCAP	—LPC array processor; an LPC voice processor developed by CHI
LPCM	—LPC microprocessor; an LPC vocoder developed by LL
LPVT	—LEXNET packet voice terminal; developed by LL
MP32	—host computer used at CHI for ARPANET packet speech
NDRE	—Norwegian Defense Research Establishment, Oslo, Norway
NVP	—network voice protocol
PCI	—packet/circuit interface; developed by LL
PDP-11	—a family of computers (programmable data processors) manufactured by DEC
PRNET	—packet radio network

PSAT	—multiprocessor packet satellite IMP developed by BBN for WB SATNET multiprocessor
PTP	—point-to-point
RFNM	—request-for-next-message; an acknowledgment message in ARPANET
SATNET	—the Atlantic packet satellite network
SCRL	—Speech Communications Research Laboratory
SF	—store-and-forward
SIMP	—satellite IMP; developed by BBN for SATNET
SIU	—speech interface units; developed by SRI
SPS-41	—a signal-processing computer developed by Signal Processing Systems, Inc.
SRI	—SRI International, Menlo Park, CA
ST	—stream protocol; an internet transport protocol for speech and other real-time traffic
STN	—switched telephone network
STNI	—STN interface; developed by ISI
T1	—standard digital carrier format used in telephony; operates at 1.544 mbits/s and carries 24 channels
TASI	—time-assigned speech interpolation; technique for saving bandwidth by transmitting only during talkspurts
TOE	—telephone office emulator; circuit switch emulator developed by LL
TX2	—host computer used at LL for early packet speech experiments
UCL	—University College, London
UMC-Z80	—a microprocessor-based input-output board used in the LL miniconcentrator gateway
VFR	—variable-frame rate; refers to vocoders operating at variable rate
WB SATNET	—the wide-band packet satellite network

Acknowledgment

The packet speech and wide-band network experiments and system developments described here were initiated by Dr. R. E. Kahn, DARPA Information Processing Techniques Offices, who has provided leadership and numerous technical contributions throughout the course of the work. Since 1978, Col. D. A. Adams, DARPA, has provided guidance, support, and technical contributions to the packet speech efforts. As noted in the text and references, the packet speech developments described here are the result of the efforts of many individuals at a number of organizations. For their contributions to the packet speech system developments and experiments, we specifically wish to cite the following individuals: D. Cohen, S. Casner, and R. Cole of ISI; E. Craighill of SRI; M. McCammon of CHI; and H. Heggestad, W. Kantrowitz, C. McElwain, and G. O'Leary of Lincoln Laboratory. We would like to acknowledge the technical contributions and cooperative efforts of these and many other colleagues who have made this paper possible.

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APPENDIX C

Network Working Group Audio-Video Transport Working Group
Request for Comments: 1889 H. Schulzrinne
Category: Standards Track GMD Fokus
S. Casner
Precept Software, Inc.
R. Frederick
Xerox Palo Alto Research Center
V. Jacobson
Lawrence Berkeley National Laboratory
January 1996

RTP: A Transport Protocol for Real-Time Applications

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Abstract

This memorandum describes RTP, the real-time transport protocol. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers.

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1. Introduction

This memorandum specifies the real-time transport protocol (RTP), which provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. Those services include payload type identification, sequence numbering, timestamping and delivery monitoring. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols (see Section 10). RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network.

Note that RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence.

While RTP is primarily designed to satisfy the needs of multi-participant multimedia conferences, it is not limited to that particular application. Storage of continuous data, interactive distributed simulation, active badge, and control and measurement applications may also find RTP applicable.

This document defines RTP, consisting of two closely-linked parts:

- o the real-time transport protocol (RTP), to carry data that has real-time properties.

- o the RTP control protocol (RTCP), to monitor the quality of service and to convey information about the participants in an on-going session. The latter aspect of RTCP may be sufficient for "loosely controlled" sessions, i.e., where there is no explicit membership control and set-up, but it is not necessarily intended to support all of an application's control communication requirements. This functionality may be fully or partially subsumed by a separate session control protocol,

which is beyond the scope of this document.

RTP represents a new style of protocol following the principles of application level framing and integrated layer processing proposed by Clark and Tennenhouse [1]. That is, RTP is intended to be malleable to provide the information required by a particular application and will often be integrated into the application processing rather than being implemented as a separate layer. RTP is a protocol framework that is deliberately not complete. This document specifies those functions expected to be common across all the applications for which RTP would be appropriate. Unlike conventional protocols in which additional functions might be accommodated by making the protocol more general or by adding an option mechanism that would require parsing, RTP is intended to be tailored through modifications and/or additions to the headers as needed. Examples are given in Sections 5.3 and 6.3.3.

Therefore, in addition to this document, a complete specification of RTP for a particular application will require one or more companion documents (see Section 12):

- o a profile specification document, which defines a set of payload type codes and their mapping to payload formats (e.g., media encodings). A profile may also define extensions or modifications to RTP that are specific to a particular class of applications. Typically an application will operate under only one profile. A profile for audio and video data may be found in the companion RFC TBD.

- o payload format specification documents, which define how a particular payload, such as an audio or video encoding, is to be carried in RTP.

A discussion of real-time services and algorithms for their implementation as well as background discussion on some of the RTP design decisions can be found in [2].

Several RTP applications, both experimental and commercial, have already been implemented from draft specifications. These applications include audio and video tools along with diagnostic tools such as traffic monitors. Users of these tools number in the thousands. However, the current Internet cannot yet support the full potential demand for real-time services. High-bandwidth services using RTP, such as video, can potentially seriously degrade the quality of service of other network services. Thus, implementors should take appropriate precautions to limit accidental bandwidth usage. Application documentation should clearly outline the limitations and possible operational impact of high-bandwidth real-

time services on the Internet and other network services.

2. RTP Use Scenarios

The following sections describe some aspects of the use of RTP. The examples were chosen to illustrate the basic operation of applications using RTP, not to limit what RTP may be used for. In these examples, RTP is carried on top of IP and UDP, and follows the conventions established by the profile for audio and video specified in the companion Internet-Draft draft-ietf-avt-profile

2.1 Simple Multicast Audio Conference

A working group of the IETF meets to discuss the latest protocol draft, using the IP multicast services of the Internet for voice communications. Through some allocation mechanism the working group chair obtains a multicast group address and pair of ports. One port is used for audio data, and the other is used for control (RTCP) packets. This address and port information is distributed to the intended participants. If privacy is desired, the data and control packets may be encrypted as specified in Section 9.1, in which case an encryption key must also be generated and distributed. The exact details of these allocation and distribution mechanisms are beyond the scope of RTP.

The audio conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration. Each chunk of audio data is preceded by an RTP header; RTP header and data are in turn contained in a UDP packet. The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet so that senders can change the encoding during a conference, for example, to accommodate a new participant that is connected through a low-bandwidth link or react to indications of network congestion.

The Internet, like other packet networks, occasionally loses and reorders packets and delays them by variable amounts of time. To cope with these impairments, the RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source, so that in this example, chunks of audio are contiguously played out the speaker every 20 ms. This timing reconstruction is performed separately for each source of RTP packets in the conference. The sequence number can also be used by the receiver to estimate how many packets are being lost.

Since members of the working group join and leave during the conference, it is useful to know who is participating at any moment and how well they are receiving the audio data. For that purpose,

APPENDIX D

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Request for Comments: 2543 ACIRI
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March 1999

SIP: Session Initiation Protocol

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Copyright Notice

IESG Note

The IESG intends to charter, in the near future, one or more working groups to produce standards for "name lookup", where such names would include electronic mail addresses and telephone numbers, and the result of such a lookup would be a list of attributes and characteristics of the user or terminal associated with the name. Groups which are in need of a "name lookup" protocol should follow the development of these new working groups rather than using SIP for this function. In addition it is anticipated that SIP will migrate towards using such protocols, and SIP implementors are advised to monitor these efforts.

Abstract

The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants. These sessions include Internet multimedia conferences, Internet telephone calls and multimedia distribution. Members in a session can communicate via multicast or via a mesh of unicast relations, or a combination of these.

SIP invitations used to create sessions carry session descriptions which allow participants to agree on a set of compatible media types. SIP supports user mobility by proxying and redirecting requests to the user's current location. Users can register their current location. SIP is not tied to any particular conference control protocol. SIP is designed to be independent of the lower-layer transport protocol and can be extended with additional capabilities.

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1 Introduction

1.1 Overview of SIP Functionality

The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots", such as a media storage service. SIP can invite parties to both unicast and multicast sessions; the initiator does not necessarily have to be a member of the session to which it is inviting. Media and participants can be added to an existing session.

SIP can be used to initiate sessions as well as invite members to sessions that have been advertised and established by other means. Sessions can be advertised using multicast protocols such as SAP, electronic mail, news groups, web pages or directories (LDAP), among others.

SIP transparently supports name mapping and redirection services, allowing the implementation of ISDN and Intelligent Network telephony subscriber services. These facilities also enable personal mobility. In the parlance of telecommunications intelligent network services, this is defined as: "Personal mobility is the ability of end users to originate and receive calls and access subscribed telecommunication services on any terminal in any location, and the ability of the network to identify end users as they move. Personal mobility is based on the use of a unique personal identity (i.e., personal number)." [1]. Personal mobility complements terminal mobility, i.e., the ability to maintain communications when moving a single end system from one subnet to another.

SIP supports five facets of establishing and terminating multimedia communications:

User location: determination of the end system to be used for communication;

User capabilities: determination of the media and media parameters to be used;

User availability: determination of the willingness of the called party to engage in communications;

Call setup: "ringing", establishment of call parameters at both called and calling party;

Call handling: including transfer and termination of calls.

SIP can also initiate multi-party calls using a multipoint control unit (MCU) or fully-meshed interconnection instead of multicast. Internet telephony gateways that connect Public Switched Telephone Network (PSTN) parties can also use SIP to set up calls between them.

SIP is designed as part of the overall IETF multimedia data and control architecture currently incorporating protocols such as RSVP (RFC 2205 [2]) for reserving network resources, the real-time transport protocol (RTP) (RFC 1889 [3]) for transporting real-time data and providing QOS feedback, the real-time streaming protocol (RTSP) (RFC 2326 [4]) for controlling delivery of streaming media, the session announcement protocol (SAP) [5] for advertising multimedia sessions via multicast and the session description protocol (SDP) (RFC 2327 [6]) for describing multimedia sessions. However, the functionality and operation of SIP does not depend on any of these protocols.

SIP can also be used in conjunction with other call setup and signaling protocols. In that mode, an end system uses SIP exchanges to determine the appropriate end system address and protocol from a given address that is protocol-independent. For example, SIP could be used to determine that the party can be reached via H.323 [7], obtain the H.245 [8] gateway and user address and then use H.225.0 [9] to establish the call.

In another example, SIP might be used to determine that the callee is reachable via the PSTN and indicate the phone number to be called, possibly suggesting an Internet-to-PSTN gateway to be used.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed, but SIP can be used to introduce conference control protocols. SIP does not allocate multicast addresses.

SIP can invite users to sessions with and without resource reservation. SIP does not reserve resources, but can convey to the invited system the information necessary to do this.

1.2 Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in RFC 2119 [10] and indicate requirement levels for compliant SIP implementations.

APPENDIX E

Fusion will help raise Net voices

Telecommunications technology provider Natural MicroSystems plans to announce Fusion, a combination of hardware and software components that it wants to license to developers of telephony applications for IP-based networks.

Tech Industry

December 3, 1996
5:30 AM PST

by CNET News staff

Telecommunications technology provider **Natural MicroSystems(NMSS)** plans next week to announce Fusion, a combination of hardware and software components that it wants to license to developers of telephony applications for IP-based networks.

The hype of Internet telephony--touted as free long distance calls through the PC desktop--has died down recently as users contemplate the difficulties, including inevitable time delays, in conducting real-time conversations over the public Internet. But the new products are aimed at corporations using private data networks that are also a viable market for the nascent technology.

Building-block tools like Fusion are designed for developers and system integrators who build applications that don't always rely on real-time conversations, such voice mail and fax service. They are also used to supplement online customer service with voice capability, like a Web-based shopping site where a human being is ready to provide voice assistance to a browsing surfer.

Lucent Technologies, **Inter-Tel**, and **Netiphone** next week will announce product plans based on Fusion.

Fusion consists of a board that interfaces with the telephone network and a board carrying an Internet protocol router. The two boards, which work together with a third board to compress incoming voice data by a factor of ten, can handle traffic on 24 analog phone lines or one T1 connection, said company officials.

Running on a Windows NT server, Fusion uses an audio compression-decompression algorithm (codec) developed under the recently ratified H.323 videoconferencing standard, which was created to deal with the problems of private local and wide area networks that tend to lose data packets. However, the accompanying software development kit will allow integrators to substitute other codecs if the client component of the application under development won't support the H.323-based algorithm.

The hardware and development kit will be available for beta by March, the company said. Once the tools are in final production, the run-time environment software will sell separately for $500 per analog port, or up to $12,000 per unit sold. However, the company will make its money on royalties from products shipped by developers, officials said. And if that's the case, Internet telephony must prove more viable in businesses in 1997 than it did at home in 1996.

Tags: Tech Industry

DISCUSS: FUSION WILL HELP RAISE NET VOICES

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The Snowden effect: Privacy is good for business

Tech companies didn't look so good when Edward Snowden revealed they were helping governments spy on average people. But the revelations have worked in the industry's favor.

CNET › Internet › The Snowden effect: Privacy is good for business

Internet

June 3, 2016
5:00 AM PDT

by Laura Hautala
@lhautala

On June 6, 2013, Edward Snowden -- holed up in a Hong Kong hotel room with two Guardian reporters and a filmmaker -- told the world about a secret surveillance program that let the US National Security Agency grab people's emails, video chats, photos and documents through some of the world's biggest tech companies.

That program was called Prism, and the journalists revealed the extent of its reach just one day after reporting that the NSA was collecting phone records in bulk from Verizon. Top-secret slides intended for NSA senior analysts -- and leaked by Snowden -- listed Apple, Google, Microsoft, Yahoo, AOL, Facebook and a video chat company called PalTalk as willing partners in the surveillance program. The public uproar was immediate, even as all of the companies denied giving the NSA unfettered access to such data.

All of the companies, except Microsoft and PalTalk, declined to discuss this story on the record.

Prism was just one of Snowden's many revelations, but its disclosure kicked off a crisis of confidence and conscience throughout the technology industry. In the three years since Snowden's initial leak, Apple, Google, Microsoft, Facebook and Yahoo have become some of the biggest advocates of consumer privacy. They've beefed up encryption and other safeguards in their products and services. A few have challenged the US government in courts -- and in the court of public opinion -- in the debate over national security and personal privacy.

"These companies are now engaged in a genuine commitment to demonstrate that they're willing to protect privacy even against the US government," says Glenn Greenwald, who broke the Snowden story while a reporter for the Guardian. "That has really altered the relationship between the US government and these tech companies, and made it much, much harder to spy."

Edward Snowden poses for a photo during an interview in an undisclosed location in December 2013 in Moscow, Russia.

Barton Gellman, Getty Images

That debate reached a crescendo early this year when Apple resisted a court order forcing it to write software that would have circumvented encryption built into an iPhone 5C used by a terrorist in San Bernardino, California. Such software "would be the equivalent of a master key, capable of opening hundreds of millions of locks -- from restaurants and banks to stores and homes," **CEO Tim Cook wrote in an open letter in February to customers**. "No reasonable person would find that acceptable."

Good for business

Since 2013, Snowden has been called everything from a whistleblower and patriot to a criminal and traitor.

That characterization seems to be fluid. Take former US Attorney General Eric Holder. He oversaw the Department of Justice when it unsealed charges against Snowden on two counts of violating the **Espionage Act of 1917** and theft of government property.

But earlier this week, Holder **told political commentator David Axelrod** he thought Snowden had performed a "public service by raising the debate that we engaged in and by the changes that we made." That said, Holder also believes Snowden should return from his self-imposed exile in Russia to stand trial for his actions.

"I think there has to be a consequence for what he has done," Holder says. "But I think in deciding what an appropriate sentence should be, I think a judge could take into account the usefulness of having had that national debate."

Holder's softening perspective shows just how much the debate colors our worldview.

Consider the tech giants' public stance on privacy, which coincidentally (or not) happens to be good for business, says Greenwald. He believes they're "petrified" of being seen as NSA collaborators and of losing customers to rivals based outside the US.

Yahoo provided the first glimpse of pushback against surveillance demands. As the public uproar began in 2013, company higher-ups immediately saw the value of telling the public another story: Yahoo had its customers' backs. They even had proof: The company had **already fought and lost** a constitutional challenge to the law that authorizes Prism's collection of user data.

In 2007, the online media portal and email service fought a court order under Section 702 of the Foreign Intelligence Surveillance Act Amendments Act that **compelled it to disclose the content of email and other communications so long as 51 percent** of the people targeted were foreign.

A week after Snowden spilled the beans on Prism, Yahoo filed a request to unseal documents from that challenge. Why the rush to go public? To make sure Yahoo's 225 million monthly email users didn't lose their trust in the company, says Chris Madsen, Yahoo's assistant general counsel.

Or put another way, to protect business. All of the other companies named as Prism participants faced the same issue.

"A failure to do that in this particular industry means a significant loss in market share," Madsen says candidly.

Battle lines

But losing customers wasn't these companies' only concern. The tech industry sincerely wants to push back, says Snowden's attorney, Ben Wizner of the American Civil Liberties Union. That's because Snowden disclosed the frightening power of the NSA's other technology efforts. These include the **Muscular program**, which exploited weak points in Yahoo's and Google's data centers to scoop up unencrypted data, and **Bullrun**, which used superfast computers to decipher encrypted emails and documents.

"There was material in the Snowden disclosures that was genuinely shocking," Wizner says. "That radicalized a lot of people in the technology community."

Encryption became the tech industry's best defense in its advocacy for consumer privacy.

Apple put itself at the vanguard of that battle, upgrading its Mac OS and iOS mobile software with stronger encryption. It also showed a very **public willingness to defy the FBI** and courts that demanded Apple create backdoors into its most important product.

SNOWDEN AND SURVEILLANCE IN AMERICA (PICTURES)





1 - **5** of **15** PREV NEXT

"When the FBI has requested data that's in our possession, we have provided it," Cook wrote in an open letter to customers on February 16. "Apple complies with valid subpoenas and search warrants....We have also made Apple engineers available to advise the FBI, and we've offered our best ideas on a number of investigative options at their disposal."

But the company won't bend on encryption, according to Cook, signaling his willingness to challenge the FBI in front of a federal judge. In March, more than **40 top tech companies signed amicus briefs** supporting Apple as it prepared to face the government in a court case that, ultimately, never took place. Then last month, Apple rehired crypto expert Jon Callas, who co-founded PGP (Pretty Good Privacy), Silent Circle and Blackphone. Callas had worked for Apple in the 1990s and again between 2009 and 2011.

Google is fighting its own encryption battle in **several undecided court cases** related to phones running its Android mobile software.

That means we can expect governments to escalate their efforts to get around encryption, says Greenwald. "It's going to be like an arms race," he says. As governments develop new tools for spying, "private companies and privacy activists [will try] to use math to build a wall of numbers, essentially, around people's communication."

That's how it should be, says Denelle Dixon-Thayer, chief legal and business officer at Mozilla, which coordinates the development of the Firefox open-source web browser.

Governments spy, she says. "It's not our job to make that easy for them."

gettyimages-459251257.jpg

Edward Snowden, seen here being interviewed in December 2013, has been living in self-imposed exile in Moscow for three years.

Barton Gellman, Getty Images

The great debate

Snowden's revelations did more than pit the tech industry against government and law enforcement, and spotlight the warring demands of personal privacy and national security.

Ironically, even unexpectedly, it also made the US government more transparent about its efforts. Less than two months after those first disclosures in 2013, the office of the Director of National Intelligence **declassified documents explaining the government's bulk collection** of US phone records.

In March 2014, President Barack Obama said that the government **should stop acquiring phone data in bulk from the phone companies**. That June, the Director of National Intelligence released its **first annual transparency report**, revealing more than 1,760 court orders to collect personal data.

In November 2015, **five public advocates**, all private attorneys with expertise in privacy law, began advising the courts on ways to minimize the impact of foreign surveillance on people in the US.

And last month, **Reuters reported Congress no longer supported draft legislation** that would have let judges force tech companies to help law enforcement crack encrypted data.

Edward Snowden 27 May
@Snowden
Congress fought the internet, and the internet won.

Anti-privacy law is dead!reuters.com/article/us-usa…
pic.twitter.com/jz1QrWn0ob

Edward Snowden
@Snowden
Follow

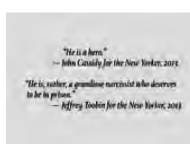
Tech companies: use this time to enable end-to-end encryption everywhere. Your future depends on making pervasive crypto a fait accompli.
7:39 PM - 27 May 2016

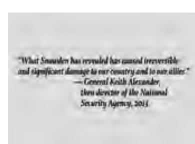
2,267 3,180

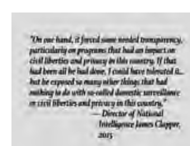
None of this means the US and other governments will end their widespread surveillance. It does, however, signal a degree of openness in telling the public how often the US goes after that data.

"Government officials have been more willing to engage in a conversation," says Margaret Nagle, Yahoo's head of US government affairs. "That has made it increasingly important that providers engage in that conversation as well."

VIEWS ON SNOWDEN: QUOTES ON LEAKS AND LIBERTY...







1 - 5 of 12 PREV NEXT

It's a beginning.

Snowden says his goal wasn't to personally end surveillance. It was to alert people that surveillance was actually happening.

"The public needs to decide whether these programs and policies are right or wrong," Snowden says in a **video published by the Guardian** in June 2013. "This is the truth.

PETITIONER APPLE INC. EX. 1009-93

This is what's happening. You should decide whether we should be doing this."

For now, the tech industry has become our proxy in that debate.

Tags: Internet, Tech Industry, Edward Snowden

DISCUSS: THE SNOWDEN EFFECT: PRIVACY IS GOOD FOR BUSINESS

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APPENDIX F

NATURAL MICROSYSTEMS LAUNCHES NEW IP TELEPHONY PLATFORM

Tuesday 27 April 1999

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Siemens selects Fusion 3.0 for its next generation of voice-over-IP solutions'

Natural MicroSystems, expert in the building blocks of standards-based telecoms solutions, today announced Fusion 3.0, its newest platform for voice and data convergence. Providing 60 ports of voice over IP in a single slot, Fusion 3.0 is a powerful, cost-effective building block for voice- and fax-over-IP platforms.

One of the first companies to use Fusion 3.0 as the basis for new IP solutions will be Siemens. The next version of Siemens's Hicom® Xpress Telephony Internet Server, a Windows NT-based gateway that enables voice and real-time fax calls over IP-based networks, will be built on Fusion 3.0.

"Partnering with Natural MicroSystems enables us to bring IP telephony solutions to market quickly and cost-effectively," said Kurt Renz, president of Siemens Enterprise Switching Networks Division, Munich. "IP telephony presents tremendous opportunities for Siemens, and we're very excited to have Natural MicroSystems as a technology partner."

Tom Valovic of IDC commented, "We estimate that the VoIP gateway equipment market will sustain triple-digit growth over the next few years. Open architectures such as Fusion 3.0 will alleviate interoperability concerns while offering an attractive price/performance model."

Bob Schechter, chairman and CEO of Natural MicroSystems, said, "Fusion 3.0 represents a breakthrough in scaleability for voice over IP, enabling our partners to bring their products to market faster and at a lower cost. Natural MicroSystems' extensive knowledge of what is required to efficiently configure data networks for voice traffic makes us the partner of choice for the world's leading telecoms companies, equipment providers and enterprise customers."

Fusion 3.0 is available in the PCI and CompactPCI form factor and supports the SPARC Solaris operating system, providing a new level of availability and reliability for IP gateways. In addition, Natural MicroSystems supports industry standards such as H.323.1, MGCP and SIP, and provides full engineering design and consulting services.

Notes to Editors

Natural MicroSystems' award-winning Fusion' is the industry's most scalable, highest-performance development platform for standards-based IP telephony solutions. Fusion enables developers to quickly create gateways with configurations from four ports to hundreds of ports without an increase in latency or a decrease in performance.

Siemens's Hicom® Xpress Telephony Internet Server's one-stage dialling allows users to access the "toll bypass" facility transparently, and the support of universal ports allows users to route voice and fax calls without having to deploy any special hardware for faxes. Furthermore, TIS 2.0 ensures that no calls are dropped due to IP data network failure or congestion via its PSTN fallback features. This features provides an automatic rerouting to a backup network in case of a failure or congestion on the IP network.

About Natural MicroSystems Natural MicroSystems (NASDAQ:NMSS), based in Framingham, Mass., is the technology leader in Open Telecommunications, providing hardware and software technologies and consulting and support services for developers of high-value telecommunications solutions. The company's state-of-the-art technology and NaturalEdge Portfolio of Services and Support enable a growing international network of OEMs, VARs, systems integrators and service providers to reduce time to market, leverage development resources, and offer truly global communications products. Natural MicroSystems products are installed in more than 40 countries worldwide.

Founded in 1983, Natural MicroSystems developed the first

PC-based telephony product to utilise digital signal processor (DSP) technology and was a leader in the creation of MVIP and H.100, the industry standards for interoperability in PC-based telephony products. The quality processes at Natural MicroSystems have earned ISO 9002 certification. More information on the company is available at http://www.nmss.com (http://www.nmss.com).

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APPENDIX G

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中 文

Cisco Systems to Acquire Selsius Systems, Inc. for $145 Million

Cisco Extends New World to PBXs and Phones

SAN JOSE, Calif. -- October 14, 1998 -- Cisco Systems, Inc. today announced it has signed a definitive agreement to acquire privately-held Selsius Systems, Inc. of Dallas, Texas. Selsius is a leading supplier of network PBX systems for high-quality telephony over IP networks.

Under the terms of the acquisition, shares of Cisco common stock and cash with an aggregate value of $145 million will be exchanged for all outstanding shares and options of Selsius. In connection with the acquisition, Cisco expects a one-time charge against after-tax earnings of between $.03 and $.06 per share for purchased in-process research and development expenses in the second quarter of fiscal 1999. The acquisition is subject to various closing conditions, including approval under the Hart-Scott-Rodino Antitrust Improvements Act.

Selsius' technology will enable Cisco to accelerate the transition from conventional, proprietary circuit-switched PBXs to multi-service, open LAN systems capable of enabling the next step in data/voice integration. This acquisition enhances Cisco's data/voice/video integration strategy by bringing open, standards-based technology to conventional PBX and telephone equipment. This technology will become an integral component in the fourth phase of Cisco's five-phase voice/data/video integration strategy.

Selsius' IP phones and call manager software will initially enable small and medium businesses and branch offices to migrate voice traffic onto packet data networks. Cisco will extend the technology to the full campus environment. Cisco will also enhance this technology to enable value-added applications such as virtual call centers and unified messaging.

"This acquisition signals the new world in PBX technology," said Mario Mazzola, Senior Vice President, Enterprise line of business. "Customers should seriously consider the network PBX technology before committing to purchase old world equipment. Cisco will demonstrate its commitment to this technology by replacing its internal conventional PBX equipment over the company's 40-building campus," he added.

Selsius was founded in 1997 as a wholly-owned subsidiary of Intecom Inc. Intecom is a wholly-owned subsidiary of Lagardere SCA, a French corporation with $11 billion in revenue in media and technology. The 51 employees led by Selsius President and CEO David Tucker will become part of Cisco's Enterprise line of business headed by Senior Vice President Mario Mazzola.

About Cisco Systems

Cisco Systems,Inc. (NASDAQ: CSCO) is the worldwide leader in networking for the Internet. News and information are available at http://www.cisco.com

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APPENDIX H

COMPANY NEWS

COMPANY NEWS; 3COM IS ACQUIRING NBX FOR $100 MILLION

Published: February 23, 1999

The 3com Corporation, a big maker of data communications equipment, said yesterday that it had agreed to acquire the NBX Corporation, a small, private company that makes telephones that communicate over data networks. 3com said it would pay about $90 million in cash and $10 million in stock for NBX, which has fewer than 90 employees. The deal represents an effort by 3com to profit from the convergence of data and telephone networks, a small but swelling trend that may allow communications consumers to reduce costs. In November, NBX, based in Andover, Mass., began selling telephones that communicate within a small or medium-sized office using standard computer networks.

APPENDIX I

nuVOICE –

Next Generation VoIP Solutions

By converging voice and data traffic into a common network

based on Nuera's packet voice technology,

competitive local exchange providers

can reduce their cost-of-ownership

and deliver high margin services.

Nuera Communications, Inc.

Draft 1.0

Wednesday, August 16, 2000

Present Situation of Local Exchange Carriers

The carrier industry has experienced double-digit growth rates during the past 4 years and the strong growth continues even today as more and more consumers subscribe telephony services. The rapidly growing subscriber base along with the dropping per-minute usage charges due to increased competition is causing concern to the service providers recently. The service operators are worried about their rising costs due to subscriber growth and shrinking margins due to new competition. These carriers need low-cost alternatives for future network evolution. The solution lies in adoption of packet telephony solutions by deploying an “All-IP Core” switching network.

Circuit vs. Packet Switching

Circuit switching formed the basis of Public Switched Telephone Network (PSTN) but now this foundation is being challenged by new packet voice technology. By now, it is a commonly accepted fact that the future belongs to packet switching, not circuit switching. Even though the transition entails major investment of effort and money, carriers are willing to undertake the change to lower costs and develop competitive advantages.

How CLECs can benefit from Packet Switching?

Basic voice service is quickly becoming a commodity. This is resulting in decreased profit margins and marginally differentiated product offerings. Packet telephony can contribute to lower operating expenses and can provide enhanced voice services.

Reduce Capital Costs

IP networks are leaner and cost less to acquire and maintain in comparison to the traditional circuit switched networks. As exemplified by the daily announcements for gigabit and terabit routers, the cost/performance curve for IP based solutions is tightly coupled to advances in high-speed silicon switching engines. This is resulting in both decreased product costs and increased switching performance.
Comparing the physical size and cost of a Lucent 5ESS or Nortel DMS 500 to a state-of-the-art terabit router quickly puts this in perspective.

Additional cost savings may be realized through the consolidation of geographically dispersed offices through a common IP core. By operating a single IP core network, the carrier stands to save significantly in terms of operating expenses associated with redundant staff and equipment. Not only is the operation of multiple dedicated networks costly and complex, an IP switching core also paves the path for integration of voice and data applications into one common network. Packet switching is the only way to aggregate various types of traffic over a single core network.

Savings from Long Distance Bypass

A majority of Competitive Local Exchange Carriers does not have their own nationwide routing network for reasons of high cost and complexity. As a result, they must rely on the incumbent long distance and international carriers for completing calls between geographically dispersed locations. In that process, the CLEC incurs a steep toll fee per call levied by the international/long distance carrier. Packet telephony offers a much lower-cost and lower-complexity alternative for the local carriers to integrate voice and data communications between their remote offices (or at least between their most important locations) thereby eliminating dependence on international/long distance carriers for call completion. The long distance bypass feature of a core IP network translates into substantial savings for the local provider.

Infrastructure Extension

Packet telephony and the associated Media Gateways and Media Gateway Controllers provide a means for the CLEC to extend and expand the investment currently sunk in circuit switched equipment. Trunk group features or functions of an installed carrier class switch may be extended to any geographic location over an IP network. A specific example would be the deployment of a small gateway into a "hi-rise" environment to provide basic CLASS type Centrex services. The incremental cost at the central switching center would be small in comparison to the increased subscriber base.

Enhanced Services

The cost savings may allow an exchange carrier to stay ahead of the ever-declining price for basic voice, but not to improve its competitive positioning. Packet telephony enables a provider to stay ahead of the pack by deploying services that are difficult or impossible when voice and data travel through separate networks. The CLEC will not only offer popular services, such as voice mail, call waiting, etc., but also a new crop of innovative services stemming from the integration of voice and data into one network. The time-to-market for new applications is reduced significantly due to the tendency of IP applications to inherit from other product development efforts and also due to a strong emphasis on open interfaces and standard protocols. Traditionally, developing PSTN applications has been a slow and costly process.

Use of open APIs from third party application providers creates unlimited service possibilities, ranging from PSTN features like pre-paid calling cards to converged network features such as "click-to-talk". The flexible nature of IP networking allow services such as:

Segment	*Feature*	*Description*
Corporate	IP VPN	Office workers use *extension dialing* to call one another through IP based Voice Private Network. Calls can be traditional telephones or through PC based soft-phones.
SOHO	Home/Office Link	Similar to the IP VPN, home office workers have essentially direct access into the corporate LAN and voice system over a broadband access service (DSL, cable, wireless).
Call Centers	Distributed Call Centers	Small call centers can be operated off a centralized corporate server and distributed via IP access solutions.

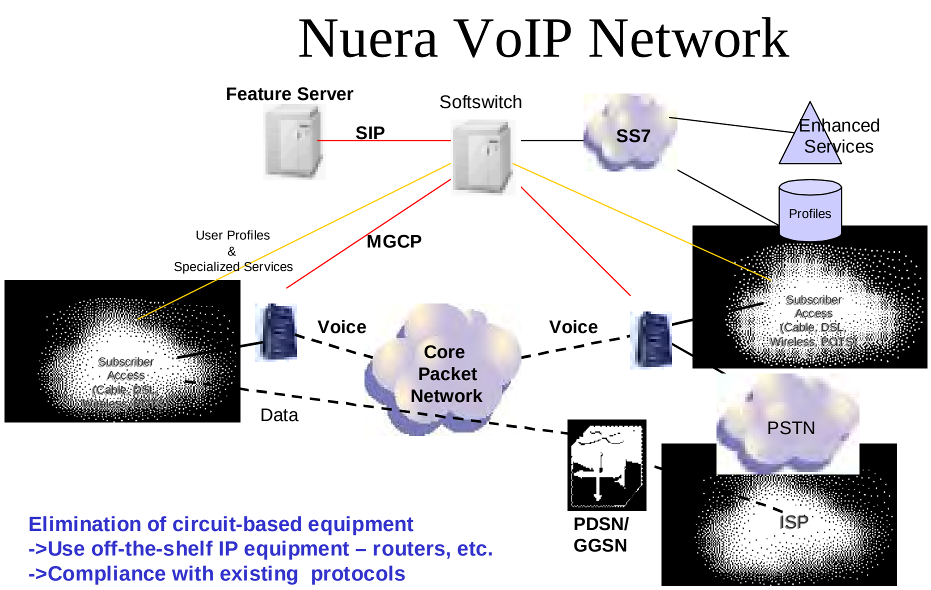
Nuera nuVOICE Solution

The local service providers expect a certain level of functionality and very high reliability from packet infrastructure solutions. The service providers wish to benefit from packet telephony by at least maintaining the current level of service expectation while simultaneously supporting high call volumes. Key requirements are:

- **N**etwork cost
- **U**ser perceived quality
- **E**ase of Management
- **R**eliability
- **A**vailability of enhanced features

CLEC Network based on Packet Voice Technology

The present generation circuit-based switching architecture performs functions, such as mobility management, subscriber profile and services management, authentication and billing. The role of the circuit-based solution will continue to regress in favor of the statistical gains delivered by packet voice and IP-based authentication, security and mobility management. The ORCA GX-21 media gateway, Nuera Softswitch (SSC) and feature servers collectively define a distributed packet-based switching center as shown below.



The Nuera solution takes into account the existing network investments made by operators and leverages existing standards for networking and telephony communication. It supports open legacy protocols such as SS7, ISDN PRI, R2 and GR303 while maintaining compatibility with evolving Voice over Packet protocols including:

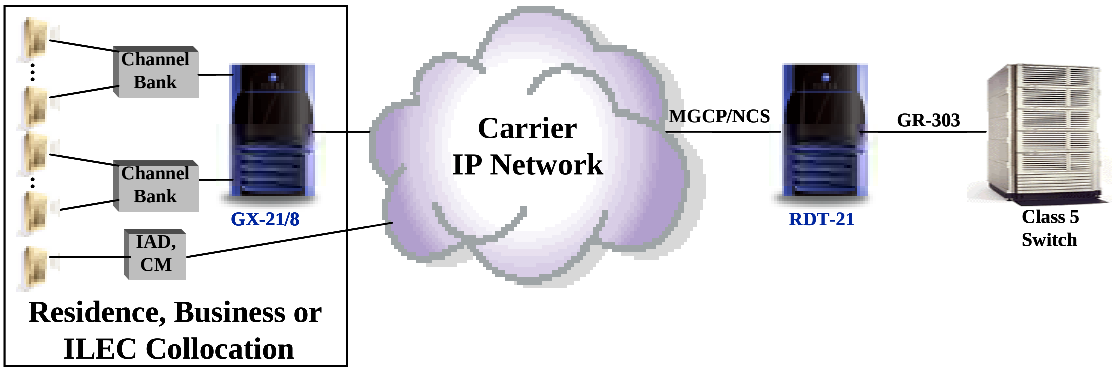
- MGCP
- NCS
- SIP
- H323

Nuera's ORCA (Open, Reliable Communications Architecture) system relies on award-winning, high-quality low-bit-rate voice, fax, and modem technology to provide a platform that dramatically lowers the cost of deploying and operating carrier-grade telephony networks. The robust network switching and routing capability of ORCA ensures that each call is switched end-to-end through the network without the need to tandem through an intermediate Class-4 switch. This reduces network complexity, operating cost, and ensures high voice quality.

The Nuera open architecture is ideally suited to carriers that want to respond quickly to their customers' changing service needs by offering innovative and differentiated services. Open database, APIs, and support of industry standards enable carriers to quickly and seamlessly integrate this powerful IP telephony platform into existing circuit-switched networks.

nuCO

Part of the nuVOICE product family, nuCO provides a cost effective VoIP-based broadband access platform.



nuCO provides a VoIP broadband access for office, campus and residential buildings. The ORCA Gateway (GX-21 or GX-8) or IAD consolidates TDM traffic to VoIP. The ORCA RDT-21 (Remote Digital Terminal) converts VoIP to GR303. The nuCO solution provides the same Class 5 features and same user experience as traditional architectures. Therefore, the nuVOICE architecture is transparent to the regular end-user.

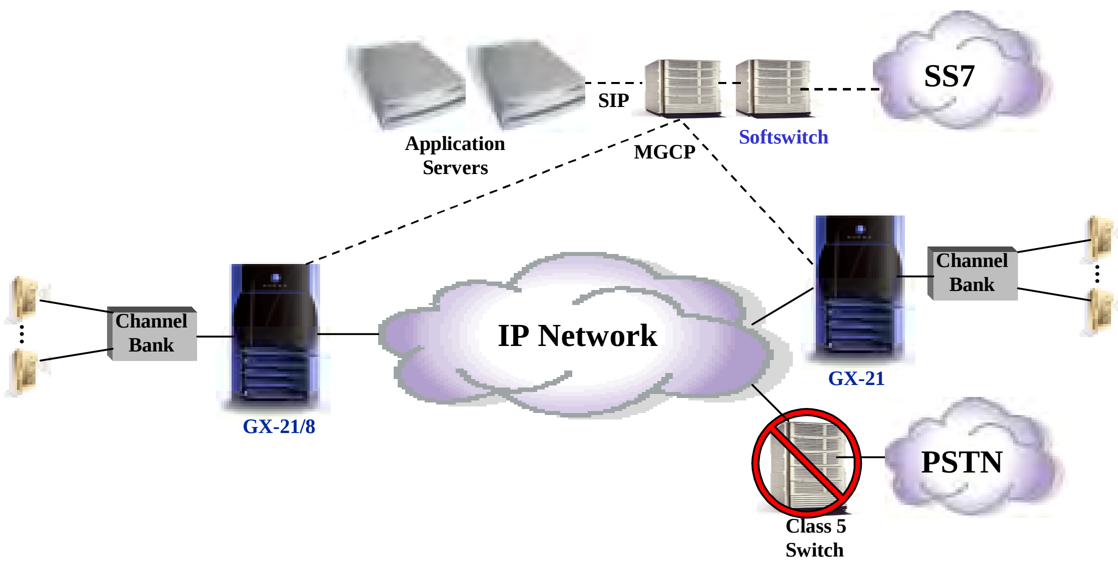
Unlike other broadband access solutions, nuCO has no medium bias. nuCO can be supported by fiber (OC-N, DWDM, HFC. . .), wireless (LMDS, WLL. . .), and copper (T1, DSL, leased lines, cable). This flexibility in the network design allows for ubiquitous service offerings over any network platform.

One of the economic advantages to nuCO is the ability to merge both the data and voice networks onto a common architecture. By consolidating voice and data, complex overlay networks and their associated overhead costs are eliminated. Also, the solution replaces costly TDM equipment including DLCs.

As a packet based architecture, nuCO can take advantage of the statistical nature of voice traffic resulting in either an increased subscriber base for a given central site configuration or smaller capital requirement for a given customer base. Both scenarios translate into lower overall costs. Over subscription of at least four subscribers per CO connection would be quite conservative. In addition, since bandwidth is only utilized when a call is active there is an inherent load balancing of bandwidth resources.

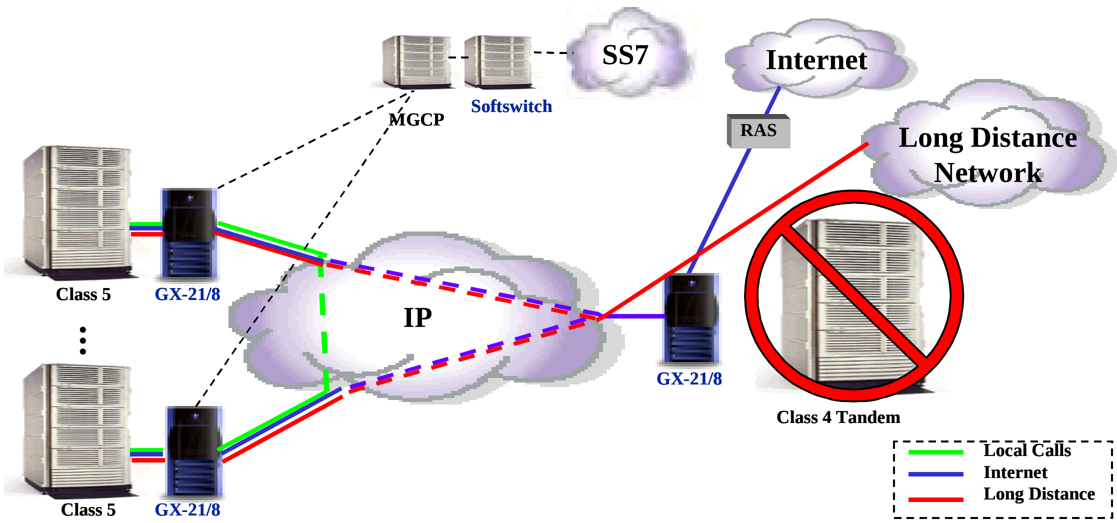
nuCO Migration to Class 5 Replacement

The Nuera Softswitch Controller (SSC) allows for future migration to an all IP network and replaces costly Class 5 switches.



RDT-21s can be redeployed as GX-21s by a simple software upgrade, thereby protecting the initial equipment investment. By streamlining the network into a full VoIP network, this prepares for future migration of next-generation services. Enhanced services such as unified messaging, “follow-me”, Web-based services, calling cards, and billing are revenue-generating services carriers can offer with very low overhead and a high payback.

nuTANDEM Migration for Class 4 Tandem Replacement



The migration path to Class 4 Tandem replacement includes tandem offloading. The ORCA Gateways allow PSTN trunks to be carried over an IP network. The data and voice network is combined and offloads traffic from the Class 4 Tandem. When the Softswitch (SSC) is introduced into the network, it offloads Internet calls from the Class 4. The SSC offsets costly switch expansions with "internet redirect" and supports SS7 networking. The ORCA SSC allows all local calls to be offloaded from the Class 4 by providing all the switching capabilities that the Class 4 would have otherwise provided. Finally, at the end of this migration path is the full replacement of the Class 4 Tandem. At that point, both local and long distance traffic can be routed over the IP network by using the nuTANDEM solution.

Network Scalability

The voice IP core can be easily scaled from a very small to a very large network. This highly scaleable architecture is particularly interesting to competitive local service providers in situations where they want to cover a small population and purchasing a full-fledged carrier class switch is not a cost-effective method. Each softswitch controls multiple media gateways and additional gateways can be added in a single softswitch control domain in an incremental fashion as traffic needs increase. Quite similarly, if a softswitch capacity is reached, it is easy to create another softswitch domain and interconnect the two domains using Session Initiation Protocol (SIP), an industry standard protocol. The feature servers can also be replicated as needed. The distributed, database-driven nature of this architecture ensures that there is no central bottleneck in the system.

Network Management

Integration of packet voice network management into a service provider's existing Operation Support System (OSS) environment is extremely important. The Nuera solution offers an Element Management System (EMS) that consolidates key management functions:

- Configuration Management
- Fault Management
- Performance Management
- Accounting Management

The EMS provides an easy Graphic User Interface (GUI) for network operators and APIs to tie the EMS to the service provider's overall network management system.

Carrier Grade Packet Telephony

The ORCA GX-21 Gateway is a carrier grade packet voice gateway designed to meet the standards of a traditional central office switch. Its salient features include:

- 99.999% Availability
- Full Redundancy and Hot Swap Capability
- High Channel Density of up 68 T1s or E1s (or 2048 voice ports) per Gateway
- Lowest packet transmission delay in industry
- NEBS-3 Compliant
- Simplified operation, administration, management, and provisioning
- Subscriber Voice Interfaces
    - Signaling: MGCP, SIP, CAS, R2, FAS (SS7, CC7, and ISDN)
    - Electrical: DSX-1, T1/SF/ESF, and CCITT G.703/704
- Packet Data Interfaces
    - LAN: 100Base-T, Serial V.35, RS422, and RS232
- Vocoder Support
    - E-CELP: 4.8, 7.4, and 9.6 Kbps
    - G.711: 64 Kbps
    - G.723.1: 5.3 and 6.3 Kbps
    - G.726: 32 and 40 Kbps
    - G.729: 8 Kbps
    - G.729A: 8 Kbps
    - GSM EFR: 12.2 Kbps
    - Real-time Fax Support: 2.4, 7.2, 9.6 and 14.4 Kbps, programmable
    - Echo Cancellation: G.165 compatible, 0-49 msec adaptive delay

Enhanced Services

An ORCA gateway in combination with a centralized softswitch provides a robust, carrier-class switching platform. One or multiple softswitches provide all of the call control and signaling services to a network of ORCA gateways. Additionally, the softswitch offers a SIP-based API to third-party systems that provide custom, enhanced network services. The ORCA gateway communicates with the softswitch using the industry-standard Media Gateway Control Protocol (MGCP). As a result, third-party softswitches can be deployed to control the network of ORCA gateways.

Market Leader

Nuera has the broadest deployable product offering in this market space including the ORCA GX-21, GX-8, the ORCA RDT-21, the ORCA SSC, and the NueraView NMS. All these components translate into cost savings and network efficiency.

Nuera has the credentials to prove its quality and reputation in the industry. Nuera has been rated "Best in Test" two years in a row by *Business Communications Review* from test reports prepared by Mier Laboratories. *Network World* recently awarded the ORCA GX-8 the highest honor of the "World Class Award." Nuera has also been the recipient of "Hot Product Awards" from *Data Communications,* and Editors' Choice and Product of the Year awards from *Internet Telephony* and *Computer Telephony* magazines.

Nuera prides itself as a leader in the interoperability initiative. Many resources are dedicated full-time to promote and test interoperability with many enhanced service providers, gateway and softswitch manufacturers, billing providers, and Class 5 switch manufacturers. As a founding member of the International Softswitch Consortium and co-founder of Voice Over IP Forum, Nuera sets the standards for the industry's adoption of MGCP and SIP.

Contact Nuera

Additional information about Nuera Communications, Inc. can be found on the Web at www.nuera.com. For more information about nuVOICE VoIP solutions, please contact:

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Fax: (858) 625-2422

APPENDIX J

Sonus Networks Gateway Switch, ip Telephony Suite Are First with the Capacity, Features and Services to Drive Global-Scale ip Telephony

For press/analyst information:
Maureen MacGregor
Pelorus Communications
978-779-8969

For investor information:
Rubin Gruber
Sonus Networks
978-692-8999 x2222

Rapid Development and Deployment of New Carrier Services Re-Defines Competitive Time-to-Market, Creates Major New Service Revenue Potential.

WESTFORD, Mass., December 15, 1998 – Sonus Networks, innovators in carrier-class IP telephony products and services, today introduced the Sonus Open Services Architecture™ (OSA), the industry's first open architecture that supports the immediate development and delivery of new telephony services. The Open Services Architecture lets both carriers and a new breed of independent applications developers rapidly deliver competitive new telephony services. Carriers and industry analysts confirm the OSA's potential to re-shape the competitive dynamics of the carrier market.

Chris Rothlis, Vice President of New Product Development at IXC, noted, "The ability to innovate and immediately deliver new products and services will be a key differentiator for new and incumbent carriers. Sonus is positioned to deliver a true carrier-class IP telephony platform."

The Sonus IP Telephony Suite and its Open Services Architecture are targeted at new converged-network carriers and incumbent carriers, both of whom need to rapidly introduce competitive new services, add massive incremental call capacity, and ensure full interoperability with and graceful migration from existing circuit networks. The Sonus IP Telephony Suite and its flagship Gateway Switch let these organizations redefine the state-of-the-art for toll quality voice and create important new customer services to fuel and expand demand. Moreover, carriers achieve substantial operational savings from employing the Sonus Suite as part of a converged IP network for voice, fax, data, video, and other applications.

THE SONUS OPEN SERVICES ARCHITECTURE SLASHES TIME-TO-MARKET OF NEW SERVICES

Sonus closely collaborated with carriers to develop the Open Services Architecture. The result is an open and unique platform that fosters innovation, slashes time-to-market for new services, and redefines the dynamics of the competitive carrier market.

ENABLING NEW THIRD PARTY DEVELOPERS AND ENTERPRISE SERVICES

The Open Services Architecture conforms to existing standards, with published APIs that open the carrier services market to a new segment of developers who can - for the first time - build new caller services on IP servers. The Open Services Architecture also allows carriers to offer services that their customers can configure and manage themselves via open interfaces such as Web browsers. Sonus Chairman Rubin Gruber observed: "The Open Services Architecture is the first true open blueprint for innovation and expanded IP Telephony services. It clearly sets Sonus apart from both traditional circuit switch vendors and IP telephony vendors who use closed development platforms."

Key Open Service Architecture elements include:

- Interworking with SS7-based services, allowing continued support for existing applications on carriers' SCPs or supplied by service providers.
- Policy-based service management, allowing centralized management of services and provisioning while distributing processing to maximize scalability and minimize cost;
- Rapid service development, allowing new services such as conferencing, language translation, enterprise VPNs, and merged multi-media to be implemented on standard hardware and software platforms;
- Open APIs, allowing equipment from multiple vendors to interact to provide the desired services. Carriers can flexibly combine services developed internally, by network equipment providers, and by third parties;

- Full Interoperabilitythrough compliance with key standards such as LDAP and the emerging MGCP;
- Script-based service definition, allowing carriers to rapidly define and deliver new services by simply using scripts that invoke processing facilities within the gateways; and

"Sonus' attention to real service provider requirements, from network management details to price/performance and the ability to develop new services independently and rapidly, distinguishes them from both legacy switch developers and IP telephony market entrants with proprietary platforms, and holds the potential to re-shape the pricing and competitive nature of the carrier market," noted Probe Research Executive Vice President Hilary Mine. "The Sonus Internet Telephony Suite's true carrier-class capacity, scalability, form factor and ability to interoperate with and enhance the existing telephony infrastructure positions Sonus as the leading vendor prepared to leverage the multi-billion market for IP Telephony," observed Paul Johnson, leading industry analyst and a co-author with Geoffrey A. Moore of The Guerilla Game: An Investor's Guide to Picking Winners in High Technology.

Sonus President and CEO Hassan Ahmed summarized: "While IP telephony will undoubtedly yield early and substantial operational savings, its most powerful impact will be in the redefinition of the market forces that define carrier competitiveness. The Open Services Architecture drives this shift, opening the door to a new market era of advanced, targeted customer services and service revenues."

AVAILABILITY

The Sonus Open Services Architecture will be delivered on the Sonus Gateway Switch (see related press release) and future Sonus IP Telephony solutions.

ABOUT SONUS NETWORKS

Sonus Networks, Inc. is developing and marketing the next generation of carrier-grade IP telephony. Its equipment facilitates the movement of telephony from traditional circuit networks to packet networks, enabling a host of new carrier and end user services. The Sonus management and engineering teams have proven success records, having led organizations such as Ascend Communications (Nasdaq:ASND) and Summa Four, recently acquired by Cisco (Nasdaq:CSCO), where they directed the development and delivery of carrier-class equipment to support data, voice and multimedia information. Sonus Networks was recently awarded the prestigious Hot Startup of the Year Award by Data Communications Magazine. Additional information is available at **www.sonunet.com**

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Sonus Networks Psx6000 Softswitch Sets Standard for Scalability, Reliability

FOR MORE INFORMATION, PLEASE CONTACT:

Sonus Networks
Beth Morrissey
978-589-8579
bmorrissey@sonusnet.com

SoftSwitch Achieves More Than 1,600 Calls Per Second in Performance Tests;
Delivers Rich Set of Features, Functionality for Call Management and Service Creation

WESTFORD, Mass., August 2, 2000 – Sonus Networks (Nasdaq: SONS), a leading provider of voice infrastructure products for the new public network, today revealed results of recent performance tests in which the company's PSX6000Ô SoftSwitch demonstrated the industry's highest levels of performance, scalability and reliability. The performance analysis and validation were conducted by Mier Communications, a leading independent networking consultancy and test center. In these benchmarks, Sonus' PSX6000 SoftSwitch achieved 1,650 calls per second, equivalent to 5.94 million BHCA (busy hour call attempts). Additionally, the PSX6000 proved to be extremely reliable, performing at high processing rates for an extended period of time without dropping a single call.

"We were extremely impressed with the performance and the scalability of the Sonus architecture," said Mike Hommer, lab testing manager, Mier Communications. "Sonus' PSX6000 is a highly reliable system with proven scalability and an impressive call-setup capability. All of these attributes are critical to carriers building out their next-generation voice infrastructures, and Sonus has demonstrated that it can handle these challenges."

Mier Communications tested Version 3.0 of the PSX6000 SoftSwitch, conducting tests that included load testing, analysis of the call-setup process itself, as well as tests to verify that call generators used in the testing behaved as they would in actual carrier environments.

Key findings and conclusions of the testing include:

- The PSX6000 SoftSwitch supported up to 1,650 calls per second with a 12-CPU configuration, and 1,400 calls per second with an eight-CPU configuration
- The PSX6000 handled an increasing call load as CPUs were added, from a four-CPU configuration to a 12-CPU setup
- Fail-over from the primary to secondary PSX6000 SoftSwitch occurred in less than one second without dropping a single call

"These test results reflect what we've experienced with the PSX6000," said Allan Van Buhler, vice president of global product development, Global Crossing. "We needed a solution that would allow us to integrate our three billion minutes of use a month onto our VoIP backbone using just a handful of softswitches. Combined with their GSX9000 Open Services Switch, the Sonus SoftSwitch delivers the kind of call processing performance and scalability that we require to deploy VoIP as a carrier-class enabling infrastructure."

Sonus' PSX6000 SoftSwitch is an integral component of the company's Open Services Architecture (OSA), a powerful voice infrastructure platform that enables carriers to deploy packet telephony for trunking, Internet offload and access applications. Additionally, the OSA provides an open approach through which new services can be created easily and implemented quickly. The PSX6000 interfaces with third-party feature servers and application servers to deliver revenue-generating enhanced services such as unified communications, single number service, Internet click-to-talk, voice portals and more.

"As carriers begin deploying next-generation voice networks, their requirements go beyond basic call processing functionality to softswitch solutions that are truly carrier-class, providing the highest levels of performance, redundancy and reliability," said Mike Hluchyj, founder and chief technology officer, Sonus Networks. "Our SoftSwitch combines extremely high call processing performance and scalability with a rich feature set, making the PSX6000 the industry's call management platform of choice."

ABOUT MIER COMMUNICATIONS

Mier Communications, founded in 1988, is an independent networking consultancy and product-test center located in Princeton Junction, NJ. The company pioneered the comparative assessment of networking hardware and software, having developed methodologies for testing products from ATM switches to network operating systems. In 1995, the company launched its "NetWORKS As Advertised" program, in which any vendor can submit its networking-related products for a comprehensive, independent assessment. MierComms also publishes special reports on important networking technologies. Call 609-275-7311 for more information on the latest report, "Getting VoIP to Work." Visit www.mier.com for more information on MierComms' full line of products and services.

ABOUT SONUS NETWORKS

Sonus Networks, Inc. is a leading provider of voice infrastructure products for the new public network. Sonus' solutions enable service providers to deploy an integrated network capable of carrying both voice and data traffic, and to deliver a range of innovative, new services. The Sonus Open Services Architecture (OSA) and award-winning Packet Telephony suite cut the time-to-market for competitive new service products, allowing carriers and third-party developers to expand marketshare and build important new revenue streams. Its highly scalable products fully interoperate with and extend the life and utility of today's public network. Sonus embodies in its management and staff decades of experience in developing carrier-class voice, data and multimedia solutions for implementation in the world's largest networks. Sonus, founded in 1997, is headquartered in Westford, Massachusetts. Additional information on Sonus is available at http://www.sonusnet.com.

This release may contain projections or other forward-looking statements regarding future events or the future financial performance of Sonus that involve risks and uncertainties. Readers are cautioned that these forward-looking statements are only predictions and may differ materially from actual future events or results. Readers are referred to Sonus' Prospectus dated May 24, 2000, filed with the SEC, which identifies important risk factors that could cause actual results to differ from those contained in the forward-looking statements.

PSX6000, GSX9000 and Open Services Architecture are trademarks of Sonus Networks. All other company and product names may be trademarks of the respective companies with which they are associated.

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Vonage Hits One Million Subscribers

Vonage Holdings announced Tuesday that it has placed one million lines of its VoIP offering into service.

Vonage Holdings announced Tuesday that it has placed one million lines of its VoIP offering into service.
The start-up firm is the U.S. leader in a rapidly growing field of companies offering paid broadband Internet telephone service. In a statement, Vonage chairman and CEO Jeffrey Cintron said: “In a very short time, Vonage has woken up a dormant televommunications industry.”

The firm, which has raised more than $400 million in financing, is said to be planning to raise $600 million **in an IPO (http://www.internetweek.com/news/170100293)**.

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- **Effective Digital B2B Collaboration: Overcoming IAM Challenges (https://webinar.informationweek.com/2116?keycode=SBX&cid=smartbox_techweb_webcast_8.500000582)**

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- **Defensive Vulnerability Pricing Guide (http://www.informationweek.com/whitepaper/security-management-and-analytics/security/defensive-vulnerability-pricing-model/376633?cid=smartbox_techweb_whitepaper_14.500002240)**

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- **[Gartner Report] 7 Best Practices for Your Big Data Analytics Projects (http://www.informationweek.com/whitepaper/big-data/big-data-analytics/gartner-report:-7-best-practices-for-your-big-data-analytics-projects/374763?cid=smartbox_techweb_report_7.300005696)**

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Attend GTEC Conference & Exhibition in Ottawa, Nov 1-3, 2016 (http://www.gtec.ca/ottawa/?_mc=sbx_x_iw_le_tsnr_gtec_x_x-promo&cid=smartbox_techweb_session_16.500166)

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- IT: A New World Order (http://www.informationweek.com/whitepaper/enterprise/information-management/a-new-world-order-of-it/376523?cid=smartbox_techweb_whitepaper_14.500002249)
- Defensive Vulnerability Pricing Guide (http://www.informationweek.com/whitepaper/security-management-and-analytics/security/defensive-vulnerability-pricing-model/376633?cid=smartbox_techweb_whitepaper_14.500002240)
- IoT: Providing Secure Connected Products (http://www.informationweek.com/whitepaper/cloud-security/cybersecurity/providing-secure-connected-products/375083?cid=smartbox_techweb_whitepaper_14.500002185)
- The Advanced Threat Hunting Guide (http://www.informationweek.com/whitepaper/endpoint-security/security/advanced-threat-hunting/372593?cid=smartbox_techweb_whitepaper_14.500002031)

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- Call Transfer
- Click-2-Call
- Call Return
- Caller ID Block
- Repeat Dialing
- International Call Block
- Ring Lists
- Call Hunt

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- Add a Line
Add multiple phone numbers to your Vonage account.
- Virtual Phone Number
Virtual phone numbers are secondary numbers from any available area code for your primary Vonage line.
- SoftPhone
Give your laptop its own phone number and get 500 local and long distance minutes included.
- Toll Free Plus
Have your own toll free number for only $4.99 per month. 100 incoming minutes included.
- Fax Service
Customers can get a dedicated fax line for only $9.99 per month.
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