**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on February 26, 2008.

Signed: /Laura M. Dean/
Laura M. Dean

Attorney Docket No.:
REBLP003/VOX-P003
First Named Inventor: KATIS et al.

**UTILITY PATENT APPLICATION TRANSMITTAL (37 CFR. § 1.53(b))**
(Continuation, Divisional or Continuation-in-part application)

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir: This is a request for filing a patent application under 37 CFR. § 1.53(b) in the name of inventors:
**Thomas E. KATIS, James T. PANTTAJA, Mary G. PANTTAJA and Matthew J. RANNEY**

For: **TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS**

Assigned to : RebelVox, LLC
This application is a ☒ Continuation ☐ Divisional ☐ Continuation-in-part of prior Application No.: 12/028,400, from which priority under 35 U.S.C. §120 is claimed. Prior Application No.: 12/028,400 claims the benefit of priority under 35 U.S.C. § 119(e) to U.S. provisional application No. 60/937,552 filed on June 28, 2007, and U.S. provisional application No. 60/999,619 filed on October 19, 2007.

The specification has been amended to claim priority from the parent application, or such amendment is included in a separate sheet.

Application Elements:

☒ **71** Pages of Specification, Claims and Abstract
☒ **33** Sheets of formal Drawings
☒ Declaration and Power of Attorney
- ☒ Newly executed
- ☐ Copy from a prior application (37 CFR 1.63(d) for a continuation or divisional). The entire disclosure of the prior application from which a copy of the declaration is

herein supplied is considered as being part of the disclosure of the accompanying application and is hereby incorporated by reference therein.

☐ Deletion of inventors. Signed statement attached deleting inventor(s) named in the prior application, see 37 CFR 1.63(d)(2) and 1.33(b).

Accompanying Application Parts:

☐ Do not publish this application. Nonpublication Request is attached.

☐ Assignment and Assignment Recordation Cover Sheet (recording fee of $40.00 enclosed)

☐ Power of Attorney

☐ 37 CFR 3.73(b) Statement by Assignee

☐ Information Disclosure Statement with Form PTO-1449 ☐ Copies of IDS Citations

☐ Preliminary Amendment (*New claims numbered after highest original claim in prior application.*)

☐ Return Receipt Postcard

☒ Other: Application Data Sheet

Extension of Time for Prior Pending Application

☐ A Petition for Extension of Time is being concurrently filed in the prior pending application. A copy of the Petition for Extension of Time is attached.

Fee Calculation (37 CFR § 1.16)

☒ Applicant is entitled to Small Entity Status under 37 C.F.R. §1.27.

| | (Col. 1) Total Claims | | (Col. 2) Claims | (Col. 3) Present Extra | Rate | Additional Fee |
|---|---|---|---|---|---|---|
| TOTAL | 22 | MINUS | 20 | = 3 | x 50 = | 100 |
| INDEP. | 1 | MINUS | 3 | = 0 | x 210 = | |
| | | | | Utility Search Fee | = | $510 |
| | | | | Utility Examination Fee | = | $210 |
| | | | | Utility Application Size Fee (for each additional 50 sheets that exceeds 100 sheets) | $260 | |
| | | | | [ ] First presentation of multiple dependent claim | $370 | |
| | | | | Basic Filing Fee under 37 C.F.R. §1.16(a) | = | $310 |
| | | | | | TOTAL | 1,130.00 |
| | | | | | SMALL ENTITY 50% FILING FEE REDUCTION (if applicable) | $565.00 |
| | | | | | Further SMALL ENTITY Filing Fee Reduction for Electronic Filing (- $80.00) | $485.00 |

☒ The Commissioner is authorized to charge any fees beyond the amount enclosed which may be required, or to credit any overpayment, to Deposit Account No. 500388 (Order No. REBLP003).

General Authorization for Petition for Extension of Time (37 CFR §1.136)

☒ Applicants hereby make and generally authorize any Petitions for Extensions of Time as may be needed for any subsequent filings. The Commissioner is also authorized to charge any extension fees under 37 CFR §1.17 as may be needed to Deposit Account No. 500388 (Order No. REBLP003).

☒ Please send correspondence to the following address:

**Customer Number 76743**

Date: February 26, 2008

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

PTO/SB/14 (08-05)
Approved for use through 07/31/2006. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

| **Application Data Sheet 37 CFR 1.76** | Attorney Docket Number | REBLP003/VOX-P003 |
|---|---|---|
| | Application Number | |
| Title of Invention | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | |

The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76.
This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.

## Secrecy Order 37 CFR 5.2

☐ Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)

## Applicant Information:

**Applicant 1** Remove

**Applicant Authority** ◉ Inventor | ○ Legal Representative under 35 U.S.C. 117 | ○ Party of Interest under 35 U.S.C. 118

| Prefix | Given Name | Middle Name | Family Name | Suffix |
|---|---|---|---|---|
| | Thomas | E. | Katis | |

**Residence Information (Select One)** ◉ US Residency ○ Non US Residency ○ Active US Military Service

| City | Jackson | State/Province | WY | Country of Residence i | US |
|---|---|---|---|---|---|

**Citizenship under 37 CFR 1.41(b)** i: US

**Mailing Address of Applicant:**

| Address 1 | 970 East Broadway, E-321 | | |
|---|---|---|---|
| Address 2 | | | |
| City | Jackson | State/Province | WY |
| Postal Code | 83001-9475 | Country i | US |

**Applicant 2** Remove

**Applicant Authority** ◉ Inventor | ○ Legal Representative under 35 U.S.C. 117 | ○ Party of Interest under 35 U.S.C. 118

| Prefix | Given Name | Middle Name | Family Name | Suffix |
|---|---|---|---|---|
| | James | T. | Panttaja | |

**Residence Information (Select One)** ◉ US Residency ○ Non US Residency ○ Active US Military Service

| City | Healdsburg | State/Province | CA | Country of Residence i | US |
|---|---|---|---|---|---|

**Citizenship under 37 CFR 1.41(b)** i: US

**Mailing Address of Applicant:**

| Address 1 | 4016 West Soda Rock Lane | | |
|---|---|---|---|
| Address 2 | | | |
| City | Healdsburg | State/Province | CA |
| Postal Code | 95448 | Country i | US |

**Applicant 3** Remove

**Applicant Authority** ◉ Inventor | ○ Legal Representative under 35 U.S.C. 117 | ○ Party of Interest under 35 U.S.C. 118

| Prefix | Given Name | Middle Name | Family Name | Suffix |
|---|---|---|---|---|
| | Mary | G. | Panttaja | |

**Residence Information (Select One)** ◉ US Residency ○ Non US Residency ○ Active US Military Service

| City | Healdsburg | State/Province | CA | Country of Residence i | US |
|---|---|---|---|---|---|

PTO/SB/14 (08-05)
Approved for use through 07/31/2006. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

| Application Data Sheet 37 CFR 1.76 | Attorney Docket Number | REBLP003/VOX-P003 |
|---|---|---|
| | Application Number | |
| Title of Invention | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | |

| | | | | | |
|---|---|---|---|---|---|
| **Citizenship under 37 CFR 1.41(b)** i | US | | | | |
| **Mailing Address of Applicant:** | | | | | |
| **Address 1** | 4016 West Soda Rock Lane | | | | |
| **Address 2** | | | | | |
| **City** | Healdsburg | **State/Province** | CA | | |
| **Postal Code** | 95448 | **Country**i | US | | |

**Applicant 4** Remove

**Applicant Authority** (•) Inventor ( ) Legal Representative under 35 U.S.C. 117 ( ) Party of Interest under 35 U.S.C. 118

| Prefix | Given Name | Middle Name | Family Name | Suffix |
|---|---|---|---|---|
| | Matthew | J. | Ranney | |

**Residence Information (Select One)** (•) US Residency ( ) Non US Residency ( ) Active US Military Service

| | | | | | |
|---|---|---|---|---|---|
| **City** | Oakland | **State/Province** | CA | **Country of Residence** i | US |
| **Citizenship under 37 CFR 1.41(b)** i | US | | | | |
| **Mailing Address of Applicant:** | | | | | |
| **Address 1** | 280 Euclid Avenue | | | | |
| **Address 2** | | | | | |
| **City** | Oakland | **State/Province** | CA | | |
| **Postal Code** | 94610 | **Country**i | US | | |

All Inventors Must Be Listed - Additional Inventor Information blocks may be generated within this form by selecting the **Add** button. Add

## Correspondence Information:

**Enter either Customer Number or complete the Correspondence Information section below.**
**For further information see 37 CFR 1.33(a).**

☐ **An Address is being provided for the correspondence Information of this application.**

| | | | |
|---|---|---|---|
| **Customer Number** | 76743 | | |
| **Email Address** | | Add Email | Remove Email |

## Application Information:

| | | | |
|---|---|---|---|
| **Title of the Invention** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | | |
| **Attorney Docket Number** | REBLP003/VOX-P003 | **Small Entity Status Claimed** | ☒ |
| **Application Type** | Nonprovisional | | |
| **Subject Matter** | Utility | | |
| **Suggested Class (if any)** | | **Sub Class (if any)** | |
| **Suggested Technology Center (if any)** | | | |
| **Total Number of Drawing Sheets (if any)** | 33 | **Suggested Figure for Publication (if any)** | |

PTO/SB/14 (08-05)
Approved for use through 07/31/2006. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

| Application Data Sheet 37 CFR 1.76 | Attorney Docket Number | REBLP003/VOX-P003 |
|---|---|---|
| | Application Number | |
| Title of Invention | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | |

| Publication Information: | |
|---|---|
| ☐ | Request Early Publication (Fee required at time of Request 37 CFR 1.219) |
| ☐ | Request Not to Publish. I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application has not been and will not be the subject of an application filed in another country, or under a multilateral agreement, that requires publication at eighteen months after filing. |

## Representative Information:

Representative information should be provided for all practitioners having a power of attorney in the application. Providing this information in the Application Data Sheet does not constitute a power of attorney in the application (see 37 CFR 1.32).
Enter either Customer Number or complete the Representative Name section below. If both sections are completed the Customer Number will be used for the Representative Information during processing.

| Please Select One: | ◉ Customer Number | ○ US Patent Practitioner | ○ US Representative (37 CFR 11.9) |
|---|---|---|---|
| Customer Number | 76743 | | |

## Domestic Priority Information:

This section allows for the applicant to claim benefit under 35 U.S.C. 119(e), 120, 121, or 365(c). Providing this information in the application data sheet constitutes the specific reference required by 35 U.S.C. 119(e) or 120, and 37 CFR 1.78(a)(2) or CFR 1.78(a)(4), and need not otherwise be made part of the specification.

| Prior Application Status | Pending | | Remove |
|---|---|---|---|
| Application Number | Continuity Type | Prior Application Number | Filing Date (YYYY-MM-DD) |
| | Continuation of | 12028400 | 2008-02-08 |
| Prior Application Status | Pending | | Remove |
| Application Number | Continuity Type | Prior Application Number | Filing Date (YYYY-MM-DD) |
| 12028400 | non provisional of | 60937552 | 2007-06-28 |
| Prior Application Status | Pending | | Remove |
| Application Number | Continuity Type | Prior Application Number | Filing Date (YYYY-MM-DD) |
| 12028400 | non provisional of | 60999619 | 2007-10-19 |

Additional Domestic Priority Data may be generated within this form by selecting the **Add** button. Add

## Foreign Priority Information:

This section allows for the applicant to claim benefit of foreign priority and to identify any prior foreign application for which priority is not claimed. Providing this information in the application data sheet constitutes the claim for priority as required by 35 U.S.C. 119(b) and 37 CFR 1.55(a).

Remove

| Application Number | Country [i] | Parent Filing Date (YYYY-MM-DD) | Priority Claimed |
|---|---|---|---|
| | | | ◉ Yes ○ No |

PTO/SB/14 (08-05)
Approved for use through 07/31/2006. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it contains a valid OMB control number.

| Application Data Sheet 37 CFR 1.76 | Attorney Docket Number | REBLP003/VOX-P003 |
|---|---|---|
| | Application Number | |
| Title of Invention | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | |

Additional Foreign Priority Data may be generated within this form by selecting the **Add** button. Add

## Assignee Information:

Providing this information in the application data sheet does not substitute for compliance with any requirement of part 3 of Title 37 of the CFR to have an assignment recorded in the Office.

**Assignee 1** Remove

If the Assignee is an Organization check here. ☒

| | | | |
|---|---|---|---|
| Organization Name | RebelVox, LLC | | |
| **Mailing Address Information:** | | | |
| **Address 1** | 512 Second Street | | |
| Address 2 | | | |
| **City** | San Francisco | **State/Province** | CA |
| **Country** [i] | US | Postal Code | 94107 |
| Phone Number | | Fax Number | |
| Email Address | | | |

Additional Assignee Data may be generated within this form by selecting the **Add** button. Add

## Signature:

A signature of the applicant or representative is required in accordance with 37 CFR 1.33 and 10.18. Please see 37 CFR 1.4(d) for the form of the signature.

| | | | | | |
|---|---|---|---|---|---|
| **Signature** | /Steve D Beyer/ | | | Date (YYYY-MM-DD) | 2008-02-26 |
| First Name | Steve | Last Name | Beyer | Registration Number | 31234 |

This collection of information is required by 37 CFR 1.76. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 23 minutes to complete, including gathering, preparing, and submitting the completed application data sheet form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

# Privacy Act Statement

The Privacy Act of 1974 (P.L. 93-579) requires that you be given certain information in connection with your submission of the attached form related to a patent application or patent. Accordingly, pursuant to the requirements of the Act, please be advised that: (1) the general authority for the collection of this information is 35 U.S.C. 2(b)(2); (2) furnishing of the information solicited is voluntary; and (3) the principal purpose for which the information is used by the U.S. Patent and Trademark Office is to process and/or examine your submission related to a patent application or patent. If you do not furnish the requested information, the U.S. Patent and Trademark Office may not be able to process and/or examine your submission, which may result in termination of proceedings or abandonment of the application or expiration of the patent.

The information provided by you in this form will be subject to the following routine uses:

1. The information on this form will be treated confidentially to the extent allowed under the Freedom of Information Act (5 U.S.C. 552) and the Privacy Act (5 U.S.C. 552a). Records from this system of records may be disclosed to the Department of Justice to determine whether the Freedom of Information Act requires disclosure of these records.

2. A record from this system of records may be disclosed, as a routine use, in the course of presenting evidence to a court, magistrate, or administrative tribunal, including disclosures to opposing counsel in the course of settlement negotiations.

3. A record in this system of records may be disclosed, as a routine use, to a Member of Congress submitting a request involving an individual, to whom the record pertains, when the individual has requested assistance from the Member with respect to the subject matter of the record.

4. A record in this system of records may be disclosed, as a routine use, to a contractor of the Agency having need for the information in order to perform a contract. Recipients of information shall be required to comply with the requirements of the Privacy Act of 1974, as amended, pursuant to 5 U.S.C. 552a(m).

5. A record related to an International Application filed under the Patent Cooperation Treaty in this system of records may be disclosed, as a routine use, to the International Bureau of the World Intellectual Property Organization, pursuant to the Patent Cooperation Treaty.

6. A record in this system of records may be disclosed, as a routine use, to another federal agency for purposes of National Security review (35 U.S.C. 181) and for review pursuant to the Atomic Energy Act (42 U.S.C. 218(c)).

7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.

8. A record from this system of records may be disclosed, as a routine use, to the public after either publication of the application pursuant to 35 U.S.C. 122(b) or issuance of a patent pursuant to 35 U.S.C. 151. Further, a record may be disclosed, subject to the limitations of 37 CFR 1.14, as a routine use, to the public if the record was filed in an application which became abandoned or in which the proceedings were terminated and which application is referenced by either a published application, an application open to public inspections or an issued patent.

9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

# TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS

## CROSS-REFERENCE TO RELATED APPLICATIONS

**[0001]** This application is a continuation of U.S. Application 12/028,400, filed on February 8, 2008 entitled "Telecommunication and Multimedia Management Method and Apparatus", which claims the benefit of U.S. Provisional Patent Application No. 60/937,552, filed June 28, 2007 entitled "Re-thinking Voice Communication" and U.S. Provisional Patent Application No. 60/999,619, filed on October 19, 2007 entitled "Telecommunication and Multimedia Management System and Method", all of which are incorporated herein by reference for all purposes.

## BACKGROUND

Field of the Invention.

**[0002]** This invention pertains to telecommunications, and more particularly, to a telecommunication and multimedia management method and apparatus that enables users to review the messages of conversations in either a live mode or a time-shifted mode and to transition the conversation back and forth between the two modes, participate in multiple conversations and to archive the messages of conversations for later review or processing.

Description of Related Art

**[0003]** The current state of voice communications suffers from inertia. In spite of automated switching, high bandwidth networks and technologies such as satellites, fiber optics, Voice over IP (VoIP), wireless and cellular networks, there has been little change in how people use telephones. One is still required to pick up the phone, dial another party, wait for a connection to be made, and then engage in a full-duplex, synchronous conversation with the dialed party. If the recipient does not answer, no connection is made, and the conversation does not take place.

**[0004]** At best, a one-way asynchronous voice message may be left if the recipient has voice mail. The process of delivering the voice mail, however, is burdensome and time consuming. The caller is required to wait for the phone on the other end to stop ringing, transition into the voice mail system, listen to a voice message greeting, and then leave the message. Current voice mail systems are also inconvenient for the recipient. The recipient has to dial a code to access their voice mail, navigate through a series of prompts, listen to any earlier received voice messages in the queue, and then finally listen to the message of the sender.

**[0005]** Another drawback with typical voice mail systems is the inability to organize or permanently archive voice messages. With some voice mail systems, a user may save a message, but it is automatically deleted after a predetermined period of time and lost forever.

**[0006]** Yet another problem with current voice mail systems is that a connection must be made between the caller and the voice mail system before a message can be left. If no connection is made, there is no way for the caller to leave a message.

**[0007]** Current telephone systems are based on relatively simplistic usage patterns: real-time live calls or disjointed voice mail messages, which are typically deleted as they are heard. These forms of voice communications do not capture the real power that can be achieved with voice communication or take advantage of the advances of network speed and bandwidth that is now available. Also, if the phone network is down, or is inaccessible, (e.g., a cell phone user is in an area of no coverage or the phone lines are down due to bad weather), no communication can take place.

**[0008]** In general, telephone based communications have not kept pace with the advances in text-based communications. Instant messaging, emailing, faxing, chat groups, and the ability to archive text messages, are all commonplace with text based communications. Other than voice mail, there are few existing tools available to manage and/or archive voice messages. In comparison, the tools currently available to manage telephone communications are primitive compared to text communications.

**[0009]** The corporate environment provides just one example of the weakness in current voice communication tools. There is currently no integrated way to manage voice communications across an organization as a corporate asset. Employees generally do not record or persistently store their phone conversations. Most business related

voice communication assets are gone as quickly as the words are spoken, with no way to manage or store the content of those conversations in any manageable form.

**[0010]** As an illustrative example, consider a sales executive at a company. During the course of a busy day, the executive may make a number of calls, and close several sales, with customers over the phone. Without the ability to organize, store, and later retrieve these conversations, there is no way for the executive to resolve potential issues that may arise, such as recalling the terms of one deal versus another, or challenging a customer who disputes the terms of a previously agreed upon sale. If this executive had the ability to easily retrieve and review conversations, these types of issues could be easily and favorably resolved.

**[0011]** Current tactical radio systems, such as those used by the military, fire, police, paramedics, rescue teams, and first responders, also suffer from a number of deficiencies. Most tactical radio communication must occur through a "live" radio connection between the sender of a message and a recipient. If there is no radio connection between the two parties, there can be no communication. Urgent messages cannot be sent if either the sender or the receiver does not have access to their radio, or a radio circuit connection is not established. Tactical communications are therefore plagued with several basic problems. There is no way (i) to guarantee the delivery of messages, (ii) for a recipient to go back and listen to a message that was not heard in real time; (iii) to control the granularity of the participants in a conversation; (iv) for the system to cope when there is a lack of signal integrity for a live conversation. If a message is not heard live, it is missed. There are no tools for either the sender or a recipient to manage, prioritize, archive and later retrieve (i.e. time-shift) the messages of a conversation that were previously sent.

**[0012]** Yet another drawback with tactical radio communication systems is that only one message can be sent at a time per channel. Consider an example of a large building fire, where multiple teams of fire fighters, police and paramedics are simultaneously rescuing victims trapped in the building, fighting the fire, providing medical aid to victims, and controlling bystanders. If each of the teams is using the same channel, communications may become crowded and chaotic. Transmissions get "stepped on" when more than one person is transmitting at the same time. Also there is no way to differentiate between high and low priority messages. A team inside the burning building fighting the fire or rescuing trapped victims should have a higher priority over

other teams, such as those controlling bystanders. If high priority messages are stepped on by lower priority messages, it could not only hamper important communications, but could endanger the lives of the fire fighters and victims in the building.

**[0013]** One possible solution to the lack of ability to prioritize messages is to use multiple channels, where each team is assigned a different channel. This solution, however, creates its own set of problems. How does the fire chief determine which channel to listen too at any point in time? How do multiple teams communicate with one another if they are all on different channels? If one team calls for urgent help, how are other teams to know if they are listening to other channels? While multiple channels can alleviate some issues, it can also cause confusion, creating more problems than if a single channel is used.

**[0014]** The lack of management tools that effectively prioritize messages, that allow multiple conversations to take place at the same time, that enable the time-shifting of messages to guarantee delivery, or that support archiving and storing conversations for later retrieval and review, all contribute to the problems associated with tactical radios. In first responder situations, such as with the military, police, and fire, effective communication tools can literally mean the difference between life and death, or the success or failure of a mission. The above burning building example is useful in illustrating just some of the issues with current tactical radio communications. Similar problems exist with the military, police, first responders and others who use tactical communications.

**[0015]** With packet-based networks, commonly used protocols include Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). UDP offers the advantage of fast delivery of data, but at the expense of completeness. Packets may be dropped in transit and not available when attempting to render the data as soon as possible at the destination. In spite of the shortcomings, UDP is the standard for Voice over Internet Protocol (VoIP) transmissions due to its speed attributes. On the other hand TCP does guarantee the delivery of perfect (i.e., an exact copy of the transmitted data) data, but at the expense of latency. All packets are delivered, regardless of how long it takes. This delay makes TCP impractical for use with “live” phone calls. Currently there are no known protocols that offer the performance advantages of both TCP and UDP, where “good enough” media can be transmitted for rendering as soon as possible, with the eventual delivery of a perfect copy of the media. Also there is no protocol that

determines how much information should be sent over the network based on the presence of recipients on the network and their intentions to render the data either live or in a time-shifted mode. In addition, other factors commonly considered, such as network latency, network degradation, packet loss, packet damage, and general bandwidth conditions, are used in determining how much data to transmit. Prior art systems, however, do not consider the presence and intentions of recipients. As a result, the default assumption is that the data is rendered by the recipient in real time. When a recipient is not going to render data immediately, these prior art systems unnecessarily use bandwidth when it is not needed, degrading the overall performance of the network.

**[0016]** For the reasons recited above, telephone, voicemail and tactical voice communications systems are inadequate. An improved voice and media communication and management system and method, and improvements in delivering voice and other media over packet-based networks, is therefore needed.

## SUMMARY OF THE INVENTION

**[0017]** The present invention is directed to a communication method for communicating on a first communication device. The method includes receiving at the first communication device media of a conversation with one or more participants over a network and storing on the first communication device the received media as the media is received over the network. The received media of the conversation may be rendered in either a near real-time mode or a time-shifted mode. The rendering of the media may be transitioned on the first communication device from the time-shifted mode to the near real-time mode by retrieving the stored media, rendering the retrieved media from storage at an increased rendering rate, ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the conversation as it is being received over the network and rendering the media of the conversation in the near real-time mode as it is being received over the network after the catch-up point.

## BRIEF DESCRIPTION OF THE DRAWINGS

**[0018]** The invention may best be understood by reference to the following description taken in conjunction with the accompanying drawings, which illustrate specific embodiments of the invention.

**[0019]** Figure 1 is a diagram of the architecture of the communication and media management system of the invention.

**[0020]** Figures 2A and 2B illustrate a block diagram of a Client running on a Device in the communication and management system of the invention.

**[0021]** Figure 3 is a block diagram of a Server used in the communication and media management system of the invention.

**[0022]** Figures 4A through 4D illustrate various embodiments of data payloads used in the communication and management system of the invention.

**[0023]** Figure 5 is a diagram illustrating data being transmitted over a shared IP network in accordance with the invention.

**[0024]** Figure 6 is a diagram illustrating data being transmitted over a circuit based network in accordance with the invention.

**[0025]** Figure 7 is a diagram illustrating data being transmitted across both a cellular network and the Internet in accordance with the invention.

**[0026]** Figures 8A through 8F are a series of flow diagrams illustrating a store and stream function of the communication and management system of the invention.

**[0027]** Figures 9A through 9C are flow diagrams illustrating the operation of a Payload Quality Manager (PQM) and Figures 9D through 9F are flow diagrams illustrating the Data Quality manager (DQM), both used by the Clients and Servers of the invention.

**[0028]** Figure 10 is an exemplary device having a graphical user interface that may be used with the system of the invention.

**[0029]** Figures 11A through 11F are diagrams illustrating multiple conversation management (MCMS) features of the invention.

**[0030]** Figures 12A through 12C are diagrams illustrating the multiple conversation management system – consecutive (MCMS-C) features of the invention.

**[0031]** Figures 13A through 13D illustrate a series of diagrams detailing the operation of the invention.

**[0032]** Figures 14A and 14B are block diagrams that illustrate the hardware used for running the Client and Server applications of the invention.

**[0033]** It should be noted that like reference numbers refer to like elements in the figures.

## DETAILED DESCRIPTION OF SPECIFIC EMBODIMENTS

**[0034]** The invention will now be described in detail with reference to various embodiments thereof as illustrated in the accompanying drawings. In the following description, specific details are set forth in order to provide a thorough understanding of the invention. It will be apparent, however, to one skilled in the art, that the invention may be practiced without using some of the implementation details set forth herein. It should also be understood that well known operations have not been described in detail in order to not unnecessarily obscure the invention.

### A. Functional Overview

**[0035]** The communication media management method and system supports new modes of engaging in voice conversations and/or managing multiple simultaneous conversations using a variety of media types, such as voice, video, text, location, sensor information, and other data. Users can engage in conversations by sending voice messages to designated recipients. Depending on preferences and priorities, the recipient(s) might participate in the conversation in real time, or they might simply be notified that the message is ready for retrieval. In the latter case, the recipient participates in the conversation in a time-shifted mode by reviewing and replying to the recorded message at their convenience.

**[0036]** Users are empowered to conduct communications in either: (i) a near-synchronous or "live" conversation, providing a user experience similar to a standard full duplex phone call; or (ii) in a series of back and forth time-delayed transmissions (i.e., time-shifted mode). Further, users engaged in a conversation can seamlessly transition from the live mode to the time-shifted mode and back again. This attribute also makes it possible for users to engage in multiple conversations, at the same time, by prioritizing and shifting between the two modes for each conversation. Two individuals using the system can therefore send recorded voice messages back and forth to each other and review the messages when convenient, or the messages can be sent at a rate where they essentially merge into a live, synchronous voice conversation. This new form of communication, for the purposes of the present application, is referred to as "Voxing"

**[0037]** When you "Vox" someone, the conversation consists of a series of discrete recorded messages, which are recorded in a number of locations, which may include the encoding device of the sender, (e.g. a phone or computer), servers on multiple transmission hops across the network, and the receiver's rendering device. Unlike a standard phone call or voice mail, the system provides the following features and advantages: (i) the conversation can transition between live and time-shifted or vice versa; (ii) the discrete messages of the conversation are semantically threaded together and archived; (iii) since the messages are recorded and are available for later retrieval, attention can be temporarily diverted from the conversation and then the conversation can be later reviewed when convenient; (iv) the conversation can be paused for seconds, minutes, hours, or even days, and can be picked up again where left off; (v) one can rejoin a conversation in progress and rapidly review missed messages and catch up to the current message (i.e., the live message); (vi) no dedicated circuit is needed for the conversation to take place, as required with conventional phone calls; and (vii) lastly, to initiate a conversation, one can simply begin transmitting to an individual or a group. If the person or persons on the other end notice that they are receiving a message, they have the option of reviewing and conducting a conversation in real time, or reviewing at a later time of their choice.

**[0038]** The communication media management system also supports new modes of optimizing the transmission of data over a network. The system actively manages the delivery of payloads to a recipient engaged in a conversation in real time when network conditions are less than ideal. For example when network conditions are poor, the system intentionally reduces the quality of the data for transmission to the point where it is "good enough" to be rendered upon receipt by the recipient, allowing the real time participation of the conversation. The system also guarantees the eventual delivery of an "exact" copy of the messages over time. The system and method therefore provides the advantages of both speed and accuracy. The utilization of network bandwidth is optimized by making tradeoffs between timeliness and media quality, using the presence and intentions of whether or not recipient(s) intend to review to the message immediately in real time, as well as measures of network latency, network degradation, packet loss or damage, and/or current bandwidth conditions.

**[0039]** It should be noted that the messages of conversations may contain voice only or voice, video and other data, such as sensor information. When the messages are

reviewed, they are listened to or visually reviewed, or a combination thereof, depending on the type of media contained in the messages. Although as of the filing of the present application, most conversations are voice only, it is intended that the communication system and method described herein broadly includes conversations including multiple media types, such as voice and video for example.

**[0040]** An improved voice and other media communication and management system and method is disclosed which provides one or more of the following features and functions:

i. enabling users to participate in multiple conversation types, including live phone calls, conference calls, voice messaging, consecutive or simultaneous communications;

ii. enabling users to review the messages of conversations in either a live mode or a time-shifted mode (voice messaging);

iii. enabling users to seamlessly transition a conversation between a synchronous "live" mode and a time shifted mode;

iv. enabling users to participate in conversations without waiting for a connection to be established with another participant or the network. This attribute allows users to begin conversations, participate in conversations, and review previously received time-shifted messages of conversations even when there is no network available, when the network is of poor quality, or other participants are unavailable;

v. enabling the system to save media payload data at the sender and, after network transmission, saving the media payload data at all receivers;

vi. enabling the system to organize messages by threading them sequentially into semantically meaningful conversations in which each message can be identified and tied to a given participant in a given conversation;

vii. enabling users to manage each conversation with a set of user controlled functions, such as reviewing "live", pausing or time shifting the conversation until it is convenient to review, replaying in a variety of modes (e.g., playing faster, catching up to live, jump to the head of the conversation) and methods for managing conversations (archiving, tagging, searching, and retrieving from archives);

viii. enabling the system to manage and share presence data with all conversation participants, including online status, intentions with respect to reviewing any given message in either the live or time-shifted mode, current attention to messages, rendering methods, and network conditions between the sender and receiver;

ix. enabling users to manage multiple conversations at the same time, where either (a) one conversation is current and all others are paused; (b) multiple conversations are rendered consecutively, such as but not limited to tactical communications; or (c) multiple conversations are active and simultaneously rendered, such as in a stock exchange or trading floor environment.

x. enabling users to store all conversations, and if desired, persistently archive them in a tangible medium, providing an asset that can be organized indexed, searched, transcribed, translated and/or reviewed as needed;

xi. enabling the system to provide real time call functionality using a best-efforts mode of message delivery at a rate "good enough" for rendering as soon as possible (similar to UDP), and the guaranteed eventual delivery of exact copies of the messages as transmitted by requesting retransmission of any missing or defective data from the originally saved perfect copy (similar to TCP); and

xii. enabling the system to optimize the utilization of network bandwidth by making tradeoffs between timeliness and media quality, using the presence and intentions of the recipient(s) (i.e., to either review the media in real-time or in a time-shifted mode), as well as measures of network latency, network degradation, packet loss or damage, and/or current bandwidth conditions.

**[0041]** In various embodiments, some or all of the numerous features and functions listed above may be implemented. It should be understood, however, that different embodiments of the invention need not incorporate all of the above listed features and functions.

## B. Glossary

**[0042]** Prior to explaining the details of the invention, it is useful to define some of the terms and acronyms used throughout the written description. This glossary of terms is organized into groups of System Components, Media, Media Management, People and Conversation Management.

### B.1. System Components

**[0043] Client**: A Client is the user application in the communication system, which includes a user interface, persistent data storage, and "Voxing" functionality. Users interact with the Client application, and the Client application manages all communications (messages and signals) and payload (Media) transfers that are

transmitted or received over a network. The Client supports encoding of media (e.g., the capturing of voice, video, or other data content) and the rendering of media and supports security, encryption and authentication as well as the optimization of the transmission of data across the network. A Client may be used by one or multiple Users (i.e., multi-tenant).

**[0044] Device**: A physical device that runs the Client application. A User may be actively logged into a single Device or multiple Devices at any given point of time. In various embodiments, a Device may be a general-purpose computer, a portable computing device, a programmable phone, a programmable radio, or any other programmable communication device.

**[0045] Servers**: A computer node on the communication network. Servers are responsible for routing Messages sent back and forth between Users over the network and the persistent storage and archiving of Media payloads. Servers provide routing, transcoding, security, encryption and authentication and the optimization of the transmission of data across the network.

### B.2. Media

**[0046] Message**: An individual unit of communication from one User to another. Each Message consists of some sort of Media, such as voice or video. Each Message is assigned certain attributes, including: (i) the User sending the message; (ii) the Conversation it belongs to; (iii) an optional or user created Importance Tag; (iv) a time stamp; and (v) the Media payload.

**[0047] Media**: Audio, video, text, position, sensor readings such as temperature, or other data.

**[0048] Conversation**: A thread of Messages (identified, persistently stored, grouped, and prioritized) between two or more Users on their Devices. Users generally participate in a Conversation using their Devices by either Reviewing Messages in real time or in a time-shifted mode, or creating and sending Messages of a Conversation as desired. When new Messages are created, they either define a new Conversation, or they are added to an existing Conversation.

**[0049] Head of a Conversation**: The most recent Message of a conversation that has been encoded by the most recent speaker. It is where a User is positioned in a

Conversation when reviewing "live" or where one jumps to if the "Jump To Live" feature is used.

**[0050] Multiple Conversation Management System or MCMS**: An application that runs as part of a Client application, which enables a User to engage in multiple Conversations using a variety of Media types. With the MCMS application, a User selects one Conversation among the multiple Conversations as current, where only the Messages of current conversation are rendered. For the selected current Conversation, the User may transition from a series of back and forth Messages in time-shifted mode to near-synchronous "live" mode, similar to a standard telephone conversation, and back again. The Messages of the non-selected Conversations are in a paused state. Messages associated with the non-selected Conversion will accumulate if others are still participating in those Conversations. The User may selectively transition the current Conversation among the multiple Conversations and Review the accumulated Messages of the selected current Conversation.

**[0051] Multiple Conversation Management System-Consecutive or MCMS-C**: Similar to MCMS, with the added feature of rendering and enabling Users to manage and participate in multiple Conversations consecutively through a hierarchical system of Priorities and time-shifting, which are automatically managed by the system. The MCMS-C application allows the Messages of consecutive Conversations to be rendered in a prioritized order, as opposed to MCMS where only the Messages of the currently selected Conversation are rendered. MCMS-C is particularly applicable in situations where it is important that the Messages of the consecutive Conversations are rendered, in the prioritized order, and/or the receipt of all Messages, even those belonging to lower priority Conversations, is more important than receiving the Messages in real-time. Examples of situations where MCMS-C may be suitable include, but are not limited to, hospitals, taxi fleet management, or tactical communications.

**[0052] Multiple Conversation Management System-Simultaneous or MCMS-S**: Similar to MCMS, with the added feature of enabling With MCMS-S, multiple Conversations are selected for simultaneous rendering, as opposed to MCMS where the Messages of only the selected current Conversation are rendered. The MCMS-S application is particularly applicable in situations where a User is listening to multiple Conversations at the same time, such as a trader listening to multiple brokers on

different exchanges and periodically sending trading requests to one or multiple of them simultaneously. MCMS-S may also be suitable for tactical communications as well.

**[0053] Priority**: The mechanism through which the system determines which Message to render next when a User is participating in MCMS-C. Priority is automatically managed by the system. A User can set default Priorities, or a predetermined set of system Priorities may be used. In the event of a conflict, where more than one Message is ready to be rendered at the same time, the system resolves the conflict at least partly based on Priority, to determine what Message to render immediately and what Message to time shift.

**[0054] Tags**: a set of attributes a User or the system may assign to a Conversation or a message, such as a topic (a company name), a directive ("action items"), a indicator ("conversation summary"), or any other label by which one might want to search or organize the data.

**[0055] Importance Tags**: A special Message attribute that enables a sender to specify when a Message is to be rendered, regardless of other Priority settings. An "urgent" Importance tag will override other Priorities for example. This feature is critical for tactical systems, though any system can be configured to use or disable this feature.

**[0056] Packet**: Any unit of binary data capable of being routed through a computer network. Each packet consists of header (meta data) and payload (media data). Includes standard packet protocols such as, but not limited to, Internet Protocol (IP), EvDO, UMTS or any other packet-based network, either radio, fiber optic, or wired.

**[0057] Header or Packet Header**: The portion of a packet that describes the packet; the meta data concerning the payload, its encoding type and destination.

**[0058] Vox packet**: A proprietary packet that enables the system and method to further refine and optimize the delivery of Messages, Media and other signaling information.

**[0059] Media Payload (or Payload)**: The actual Media portion of a Packet.

### B.3. Media Management

**[0060] Time Shift Delay (TSD)**: The amount of time between the arrival of a Vox Packet and the rendering of the Packet on a Device. The TSD must exceed the **Minimum Time Shift Delay**. The TSD is typically determined by the User's behavior in choosing to review the Messages of a Conversation some time after receipt.

**[0061] Minimum Time Shift Delay (MTSD)**: The time shift delay enforced by the Client to allow for jitter processing using jitter buffer techniques. This causes the system to delay rendering until an adequate number of the packets have arrived to create a usable media stream. The system will typically adaptively adjust the MTSD over time to compensate for variable conditions in the network.

**[0062] Rendering**: Delivering a Media stream to a User in a form suitable for User consumption (e.g., voice, text, graphic display, video, or a combination thereof).

**[0063] Mixing**: The Rendering of one or more Media streams. For example, the Media stream from two Participants of a Conversation may be Mixed when Rendered, creating a User experience similar to a conversation where multiple people are speaking at the same time.

**[0064] Encoding**: The process of translating Media either created by a User (such as voice or video) or otherwise originating on a Device (such as GPS or other sensor data), and converting the media into digital data to be processed by a Client.

**[0065] Adaptive Jitter Buffer**: Jitter buffers or de-jitter buffers are used to counter jitter (i.e., either the arrival of out of sequence packets or the delayed arrival of packets) introduced by packet switched networks, so that the continuous rendering of audio (or video) signals transmitted over a network can be performed without disruption. The data is stored in a buffer before Rendering to allow a reasonably sized buffer of Media to arrive. The Media may be rendered before all the Packets are received, trading off quality for currency. An Adaptive Jitter Buffer is capable of dynamically changing its size to optimize the delay/quality tradeoff.

**[0066] Persistent Infinite Message Buffer (PIMB)**: The PIMB is a storage management system for the storage of time-based Media that performs both the de-jittering of "live" data and the storage and retrieval of archive data. The PIMB further includes the additional attributes of potentially infinite and persistence storage of Media. The PIMB maintains "exact" or full copies of Vox Packets of a Message and Conversations at some or all Participant Devices and/or Servers.

**[0067] Packet Loss Compensation or Concealment) (PLC)**: During Rendering of a Media stream, the PLC component compensates for missing Packets, interpolating the results to present the stream to a reviewer. Missing Packets may be rendered as silence, or information from adjacent Packets may be used to present an interpolated sound or

image. The particular method to be used will be dependent on the media, Codecs in use, and other generally known parameters.

### B.4. People

**[0068] User**: A person who is authorized to use the system.

**[0069] Contact**: A record of either a User or non-user of the system. Users typically engage in Conversations with members on their list of Contacts. A non-user is a user that accesses or uses the system using a legacy phone, radio or other non-Client 12 enabled device.

**[0070] Group**: The association of multiple Contacts. Contacts may be selectively added or deleted from a Group. When a Conversation takes place among a Group, all the members of the Group may or may not participate.

**[0071] Channel**: Typically used for tactical communication systems. A Channel is similar to a Group in that it associates multiple Contacts with the Channel.

**[0072] Participant**: A person who is identified as a member of a Conversation. Could be a User or a non-User participant.

### B.5. Conversation Management

**[0073] Time Shifting**: Time shifting is the ability to play any Message at any time after it has been received as determined by the User-recipient. By Time-Shifting, a User may Review a Message: (i) immediately on demand by Rendering immediately after the MTSD; or (ii) time-shifted in a mode of reviewing the Message upon the discretion of the User; (iii) from the archive for searching, reconstructions, etc. of old Conversations; (iv) after a delayed period of time to accommodate the Reviewing of other higher Priority Messages (or Conversations) that need to reviewed first; (v) and/or repeatedly if necessary for the Message to be reheard and understood. In other words, Time Shifting is the ability of a user to render a Message at any time after the system imposed MTSD.

**[0074] Reviewing**: Listening, viewing, reading or otherwise observing the Media content in Messages. Reviewing may take place in either a near synchronous real-time "live mode" or the time-shifted mode.

**[0075] Intention**: Either (i) a User-defined attribute capturing whether the User wants to Review the Messages of a Conversation either as soon as possible or Review the

Messages in a time-shifted mode; (ii) implied by a User's behavior; or a combination of (i) and (ii). .

**[0076] Attention**: A user attribute capturing whether the User is Reviewing the Messages of a given Conversation at the moment.

**[0077] Catch Up To Live (CTL)**: A rendering mode that allows a User, who is not at the Head of a Conversation, to Review previous Messages more quickly to "Catch Up To Live" (i.e., the Head of the Conversation). The CTL feature may use any of a number of catch up techniques, such as the faster replay of Messages, the removal of gaps in the Media of the Messages, removal of hesitation particles, etc. When the User has caught up to live, the system seamlessly flows into the live Conversation. This is a very useful feature with conference calls, for example, in situations where a User needs to temporarily shift their attention away from the Conversation, but wishes to hear the entire Conversation upon their return.

**[0078] Catch Up Mode**: A user-configured or pre-configured mode that determines how the CTL process will catch-up (i.e., replay faster, remove silence, and hesitation particles, or a combination thereof).

**[0079] Jump To Live (JTL)**: This feature allows a user to jump from their current position to the Head of a Conversation. A user will typically use the JTL feature when they do not want to Review all of the Messages between their current position in the Conversation and the Head. When the JTL feature is implemented, the user skips over any intervening Messages and starts Rendering the "live" Messages at the head of the Conversation.

**[0080] MCMS Participant Attributes**: A set of attributes, either defined by a User, interpreted by the system from the User's behaviors, assigned by an administrator, or a combination thereof, which define the Intention, Attention, Priority, and rendering preferences of a receiver for a given Conversation. The attributes include, but are not limited to: (i) the Intention of when a receiver would like to render to the Messages of the Conversation. Possible Intention values include: "now", "time-shifted", "Catch Up To Live" (CTL), "paused", and "never"; (ii) Catch Up Mode, which is a configuration setting which determines how the CTL process should catch the receiver up to live (e.g., play faster, skip silence gaps or hesitations, or play at normal speed); (iii) Time Shift Delay (TSD), which defines how far the receiver's current position in the conversation is

from the Head of the Conversation, and (iv) the Priority of the Message with regard to the receiver's other Conversations.

## C. System Architecture

**[0081]** Referring to Figure 1, a block diagram of the telecommunication and media management system according to one embodiment of the invention is shown. The system 10 includes a plurality of Clients $12_1$ through $12_n$, running on Devices $13_1$ through $13_n$ respectively. The Devices 13 communicate with one another over a communication services network 14, including one or more Servers 16. One or more networks $18_1$ through $18_n$, is provided to couple the plurality of Devices $13_1$ through $13_n$ to the communication services network 14. In various embodiments, the networks 18 may be the Public Switched Telephone Network (PSTN), a cellular network based on CDMA or GSM for example, the Internet, a tactical radio network, or any other communication network, or a combination thereof. The communication services network 14 is a network layer on top of or otherwise in communication with the various networks $18_1$ through $18_n$. In various embodiments, the network layer 14 is either heterogeneous or homogeneous. Clients $12_1$ through $12_n$ communicate with one another and with Servers 16 over the networks $18_1$ through $18_n$ and network 14 using individual message units referred to as "Vox packets", which are described in detail below.

## D. Client Architecture

**[0082]** Referring to Figures 2A and 2B, a block diagram of a Client 12 running on a Device 13 is illustrated. As illustrated in Figure 2A, the Client 12 includes Multiple Conversation Management System (MCMS) application 20, a rendering and encoding module 21, and an MCMS applications database 22. As illustrated in Figure 2B, the Client 12 further includes a Store and Stream (SAS) module 24 with a Persistent Infinite Message Buffer (PIMB) reader 26, a PIMB writer 28, PIMB database 30, a data and network quality (DNQS) store 32, and Media driver and encoder hardware 34. The MCMS application 20 and the Store and Stream module 24 communicate with one another through message handling modules 25a and 25b respectively. The Client 12 further includes an authentication-encryption-security module 40 and a communication protocol module 44.

**[0083]** The module 40 provides authentication, encryption and security services during the transmission and receipt of "Vox" packets to and from the Client 12. The

communication protocol module 44 encapsulates Vox packets into the native packets used by the underlying network 18 connected to the Device 13 running the Client 12 when transmitting data and de-encapsulating Vox packets from the native packets when receiving data. With the modules 40 and 44, multi-party end-to-end authentication, encryption and security is provided between Clients 12. Messages are authenticated, encrypted and secured across the networks $18_1$ through $18_n$ and network 14, from a first sending Device 13 to second receiving Device 13.

### D.1.1. The MCMS Database

**[0084]** The database 22 stores and manages the persistent meta data for a number of entities in the system 10, including Contacts and Participants, Conversations and Messages (live and stored), and default Priorities, and information regarding the Servers 16. In addition, the MCMS database 22 stores the moment-to-moment operational data of a User's Conversations, presence, and status, as well as that of all the Participants conversing with the User or on the User's Contact list. For example, with regard to Conversations and Messages, the database 22 keeps track of status information, such as what Messages of a Conversation a User has or has not Reviewed, Priorities, and Catch Up To Live status for each Conversation in which the Client 12 is a Participant, the presence and status of all Participants, and other network and other system management data.

### D.1.2. The MCMS Application

**[0085]** MCMS application 20 supports the different Voxing modes of engaging in conversations and/or managing multiple conversations using a variety of media and data types (voice, video, text, location, data, etc.). Users engage in Conversations by sending Messages to designated recipients using their Client 12 enabled Devices 13. Depending on preferences and Priorities, the recipient might Review the Message in real time, or they might simply be notified that the Message is ready for Reviewing. Users can transition from a series of back and forth Messages, which are Reviewed in a time-shifted (or voice messaging) mode or in a near synchronous, full duplex conversation (similar to standard "live" phone calls) and then back to voice messaging again. The MCMS application 20 allows a User to control their interactions with their most important Conversations in real-time without missing any Messages in other ongoing Conversations. For example, the MCMS application 20 notifies a User of urgent or high

priority communications from a Conversation that they are not currently Reviewing. MCMS application 20 also enables all Messages from all Conversations to be stored for later retrieval so they can be reviewed at any time.

**[0086]** In accordance with various embodiments, there are several different operational modes of the MCMS application 20, including MCMS-Consecutive (MCMS-C) and MCMS-Simultaneous (MCMS-S), which support the consecutive and simultaneous rendering of Messages respectively. Each of these embodiments is described in more detail below. Unless specifically specified, the term "MCMS" is intended to generally mean the MCMS application 20, which includes the aforementioned different modes.

**[0087]** The MCMS application 20 is a multi-tiered architecture that includes a number of modules and services. The modules and services include the MCMS Database Module 20a, the SAS Services Module 20b, the Messaging and Signaling Services Module 20c, the User Interface Application Programming Interface (API) 20d, the User Interface Module 20e, the Conversations/Messages Management Services 20f, the Priorities Services 20g, the Contacts Service 20h, the Presence/Status Services 20i, and the Messages/Signals Services 20j.

#### D.1.2.1 The MCMS Database Module

**[0088]** The MCMS database module 20a is a service module that manages all function calls necessary for the MCMS application 20 to access the MCMS database 22.

#### D.1.2.2 The SAS Services Module

**[0089]** The SAS Services module 20b includes a set of function calls that enable communication and coordination between the MCMS application 20 and the Store and Stream module 24, and which are passed back and forth through the message handling modules 25a and 25b respectively. The set of function calls enable both the MCMS application 20 and the Store and Stream module 24 to operate as necessary to implement the various Voxing functions when invoked by Users and/or as dictated by network conditions. Some of the functionality performed by the SAS Services module 20b includes maintaining and communicating the status of Message transmissions and Message acknowledgments, the instructions for rendering of Messages, and the status and presence of Users.

#### D.1.2.3 The Messaging and Signaling Services Module

**[0090]** The Messaging and Signaling Services module 20c runs on both Clients 12 and Servers 16 and enables communication between the Client 12 and the Servers 16 of the system 10. This communication, which includes messages, data and other signals, allows the Client 12 and the system 10 to track and administer communications, network status, Users, and User status. Types of messages and signals sent between the Message and Signaling Services modules 20c running on the Clients 12 and the Servers 16 include, for example, network availability of Users, tracking of Messages that the Server 16 has sent to the Client 12 (possibly including a "high water mark") to determine if an entire message or some portion of a message is missing, (e.g., a sequence number per Participant per Conversation created by the "generating" Client), whether a user is speaking or Reviewing Messages of a given Conversation, where a User is with respect to the Head of a Conversation, or when a Participant is no longer Reviewing a Conversation live. These are examples a few of the many types of messages and signals sent between the Message and Signaling Services modules on the Clients 12 and Servers 16 and in no way should be construed as limiting the invention.

#### D.1.2.4 The User Interface API

**[0091]** The User Interface API 20d is a module that defines a set of function calls that define the programming interface between the User Interface module 20e and the underlying services of the MCMS application 20. The User Interface API 20d supports general-purpose methods such as UI application support, and all function calls necessary for a User Interface to operate the MCMS Application 20. In various embodiments, the User Interface API 20d enables the Client 12 to support a wide variety of user interfaces and device types, such as Adobe Flash-based and/or Microsoft Windows applications, cellular or mobile phone devices, PSTN devices driven with tones, a voice user interface (VUI), and physical radio communication interfaces. In various embodiments, the User Interface API module 20d enables the design of both highly flexible and highly constrained user interfaces to support the functionality of the MCMS application 20.

#### D.1.2.5 The MCMS User Interface Module

**[0092]** The MCMS User Interface module 20e supports the operation and functions of the audio and video user interface of the Client 12. The User Interface module 20e

supports a host of user interactions and can be implemented with various interaction mediums, such as, an array of graphical user interface screens, an Audio/DTMF interface, or voice user interface on the Device 13, all of which enable a User to interface with the system 10. A partial list of User interactions that are supported include, for example, functions to: log-in; manage, join, and monitor Conversations; control Conversation rendering; manage Priorities; and requests to review archived Conversations. It should be noted that this list is exemplary and in no way should be construed as limiting the invention.

#### D.1.2.6 Conversation/Message Management Services

**[0093]** The Conversation/Message management services 20f is a module which defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for the User to manage the receipt and Review of transmitted and received Media (e.g., voice or video content Messages) between the participants of a Conversation. The Messages are organized into Conversations. Media that is sent or received by the Device 13 running the application 12 is available for immediate Review while being received. The received Media is also recorded for Review in a time-shifted mode, Conversation management, and archival purposes. In an alternative embodiment, Messages or Conversations can be optionally marked for transience, specifying their desired retention requirements (e.g., some Messages will not be retained or stored beyond the requirements for immediate rendering). In yet another embodiment, Media can be optionally marked for review in a time-shifted mode only and cannot be reviewed immediately upon receipt.

**[0094]** The Conversation/Message management services 20f further enables, for each current or ongoing Conversation of the User, the sending of Media to a receiving Client 12 at any time, and the receiving Client 12 seamlessly associates these Messages with the appropriate Conversation, regardless of the actions or inaction of the receiver.

**[0095]** With the Conversation/Message management services 20f, all Conversations are essentially asynchronous. If two Users are actively engaged in a given Conversation and the User controlled delay between transmissions is minimal, the experience will be one of a synchronous full duplex conversation, as with current telephone or VoIP conversations. If either User delays their participation, for whatever reason, the Conversation drifts towards an asynchronous voice (or other Media) messaging

experience. In alternative embodiments, Conversations can be optionally Tagged as asynchronous Messages only or synchronous Messages only. In either of these cases, the Conversation cannot drift between the two modes, unless the Tag is reset. After the Tag is reset, the Conversation again may flow between near synchronous (i.e. live or real-time) and asynchronous (i.e., time-shifted or voice messaging) modes.

**[0096]** The Conversation/Message management service 20f processes the transmission and receipt of Messages in a progressive fashion. When transmitting, Media may be created while Messages are simultaneously encoded, stored and transmitted. In other words, the transmission of Messages may occur simultaneously with the generation of Media by the User (i.e., while speaking into their Device 13 or generating video). On the receiving side, the receipt, storage, and Rendering of Messages also all occur progressively. Messages do not need to be completely received before they can be Rendered. The Rendering of Messages may occur at the same time Messages are being delivered, right up to the MTSD. Further, the service 20f is also capable of the simultaneous transmission of outgoing Messages and Rendering of incoming Messages. The progressive nature of the service 20f allows a User to be engaged in a live Conversation while storing and streaming the media of the Conversation for later retrieval and review as well other functions described herein.

**[0097]** The time-shifting of Messages by the Conversation/Message management services 20f allows a User to "Catch Up To Live" on a Conversation if they missed earlier Messages or were involved in another Conversation. This time-shifting process eliminates the need for Users to broadcast a request to their entire Group or Channel to have Messages repeated. Older Messages may be replayed at any time at potentially higher speeds to save time. Users may easily skip forward and backward through their Messages and within individual Messages. The Reviewing process may be configured on a Message-Priority basis to potentially skip lower priority Messages.

**[0098]** In one embodiment, the Conversation/Message management service 20f also identifies Messages by a specific Participant (speaker) and, by default, mixes Messages of a Conversation that were delivered at the same time (MCMS-S). In an optional embodiment, a User could Review the transmissions of different Participant speakers of a Conversation separately.

**[0100]** The Conversation/Message management module 20f further allows Conversation sharing among Participants, who can be added to an active or archived Conversation. In one embodiment, an added Participant to a Conversation is provided access to the Conversation and has the ability to retrieve the previous Messages of the Conversation for Review. In an alternative embodiment, the added Participant is provided access to the Messages of the Conversation only from the point where the new Participant joined, and not any previous Messages of the Conversation.

**[0101]** The Conversation/Message management module 20f is also responsible for managing the functions used to control all rendering tasks performed by the Store and Stream module 24. These tasks include rendering Media (i.e., voice, video, etc.) appropriately for the Device 13 running application 12. These rendering tasks include, but are not limited to, the rendering of Mixed Messages (i.e., overlapping messages), as well as rendering according to user-defined criteria, such as playing faster, catching up to live, removing silence, removing hesitation particles, frequency shifting, and the ability to apply independent gain control to individual senders in a multi-party conversation.

#### D.1.2.7 Priority Services

**[0102]** The Priority service 20g is a module that defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for a User to manage the Priority of the consecutive Conversations (i.e., MCMS-C) in which the User is engaged. When a User participates in a number of consecutive live Conversations, the User is required to prioritize the Conversations. Issues arise when Messages of different Conversations are ready to be rendered at the same time. An algorithm is used to determine the order in which the Messages are rendered that considers the availability of Messages to be rendered and the Priorities set by the User. The algorithm determines that the available Messages with the highest priority are rendered first while any concurrently available Messages are time shifted automatically just enough to allow for the rendering of the higher priority Message. As rendering time becomes available, the system will automatically render the time-shifted messages according to the User's Priorities.

### D.1.2.8 Contacts Services

**[0103]** The Contacts services 20h is a module that defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for authenticating and associating one or more Contacts with Conversations. When sending a Message as part of a Conversation that is associated with a number of Contacts, all of the Contacts receive the Message.

### D.1.2.9 Presence/Status Services

**[0104]** The Presence/Status services 20i is a module that defines a set of functions that maintain the data structures and processes responsible for managing and sharing presence and status information between certain Users and/or non-users of the system. In various embodiments, the presence and status information is maintained for all User and non-users engaged in the Conversations the User of the Client 12 is engaged in, all Users and non-users in the Contacts list, or Users within a predefined domain (e.g., the members of a corporation or other organization). These examples are merely illustrative and should not be construed as limiting. The Presence/Status services 20i module may manage and share presence and status information on any defined set of Users and/or non-users.

**[0105]** The Presence/Status service 20i enables Users to monitor the status of other User's Intentions, Attention, and their Time Shift delay on any given Conversation (i.e., how far back they are from Reviewing the Messages of the Conversation live). In one embodiment, privacy controls are provided concerning availability of Presence and Status data. The Presence/Status module 20i further controls the data that enables the system 10 to deliver Messages that match the behavior and Intentions of Users. For example, a User may indicate their Status by designating an Intention to either Review or not Review a Conversation live. In response, the Presence/Status services 20i issues commands that cause the rendering of Messages either "live" or time-shifted, in accordance with the Intention of the User. In addition, the Intentions of Users are shared with the other Participants of the Conversation. The service 20i is also capable of inferring other Status values from the User's behaviors. The Presence and Status information is also used to optimize network traffic and bandwidth, as described in more detail below.

#### D.1.2.10 Messages/Signals Services

**[0106]** The Messages/Signals services 20j is a module that defines a set of functions that manage the data structures and processes responsible for messaging and signaling Users of the system 10, using special messages or audible tones. The special messages or tones may include for example an indication if a Message or Messages are live or time-shifted, whom the Message(s) are from, Priority, and other factors. The Message/Signal service 20j further has the ability to (i) signal the presence or absence of Users on the network, as well as the ability to notify if one or more Users are no longer actively Reviewing the Messages of a Conversation; (ii) "ring" or otherwise notify another User to get their attention when they are paying attention to another Conversation or not paying attention to their Device 13 at all; (iii) leave a Message for Users currently not on the network 18 to immediately review the Message the next time the individual re-connects to the network 18; (iv) generate an audible or visible feedback that alerts the sender that a sent Message was not received by recipient(s), generate a confirmation when the Message has been received by the recipient(s), and/or a confirmation indicating when the Message has been Listened to by the recipient(s); and (v) implement a Priority scheme where individuals on a Conference or tactical call may be notified that their attention is immediately needed on the call. This indication may convey multiple levels of urgency and an acknowledgement of some kind by the recipient.

#### D.1.2.11 Rendering and Encoding

**[0107]** The rendering and encoding module 21 is responsible for performing all rendering tasks for the MCMS application 20. These tasks include rendering Media appropriately for the device 13 running application 12.

### D.2 The Store and Stream Module

**[0108]** The Store and Stream module 24 supports a number of functions and performance attributes, which are described below.

**[0109]** With the Store and Stream module 24, Message transmission is essentially "full-duplex", enabling any party to send a Message at any time, even while another party is also sending a Message, or if the other party is unavailable or otherwise engaged. The Store and Stream module is able to render messages as in a live PSTN or VoIP call or

deliver them for time shifted messaging modes. It is able to optimize transmission and control Rendering according to the desires of the User.

**[0110]** The Store and Stream module 24 maintains connectivity with all target recipients (e.g., Servers 16 or other Devices 13) on the underlying network 18, manages all message, signal, and media transmissions, and optimizes the delivery speed and bandwidth usage across the network 18 to meet a User's immediate performance requirements, while managing network quality and capacity. The module 24 adapts and optimizes Media delivery commensurate with the quality and capacity of the underlying network 18. When insufficient underlying network resources are available, the quality of the transmitted Media streams can be degraded. As bandwidth becomes available, the quality of the transmitted Media streams may be increased. In addition to tradeoffs of Media quality, the Store and Stream functionality can make tradeoffs in the amount of Media transmitted in each packet based on Users' intentions to render data in real time as described below.

**[0111]** By dynamically controlling the delivery rate of Media based on the conditions of the underlying network 18, the Store and Stream module 24 is optimized to deliver time-sensitive Media that is "good enough" to Render upon receipt, and the guarantee eventual delivery of exact or full copies of the Media for archival purposes through a background process of requesting retransmission of missing, low quality, or damaged packets. As long as sufficient network resources exist to meet minimum Media quality levels, this retransmission does not impede the Rendering of live call Media. The Clients 12 of the system 10 are thus designed to bridge the performance gap between the delivery of an exact or complete copy of the Media at the expense of substantial potential latency versus the quick delivery of Media, but with no guarantees of completeness. In the context of this application, the term "good enough" means that the quality of the Media is sufficient so that when it is rendered, it is intelligible. The notion of "good enough" is therefore subjective and should not be construed in absolute terms. For example, the quality level of certain Media to be good enough may vary depending on the type of Media, circumstances, and other factors.

**[0112]** The Store and Stream module 24 further persistently stores all Media created by or otherwise originating using a Device 13 or received over the network 18 from other Device 13 and/or users. There are several significant advantages of storing this Media on the Device 13 running the Client 12: (i) it enables Users to leave a Message for

another party, even when the sender and/or the recipient has either unavailable or poor network connectivity. In the case of insufficient bandwidth, the Message will be transmitted as fast as available bandwidth can be effectively used. In the case of no connectivity, the Message is queued for transmission as soon as network connectivity becomes available, resulting in a time-shifted delivery; (ii) the User has the ability to pause, replay, fast-forward, and Catch-Up-To-Live with an ongoing Conversation, as well as retrieve and review the archived Messages of previous Conversations; and (iii) it enables the optimization of data payloads over the system 10 and improves system resilience against network bandwidth and connectivity problems that may occur from time to time.

**[0113]** The Store and Stream module 24 is also responsible for: Mixing Messages as appropriate to create overlapping Messages (generated by the normal overlap of speakers in a Conversation or background noise), simulating an actual Conversation where multiple parties are speaking; rendering transcriptions or translations of audio media; adjusting the rendering of Media according to a number of User-defined criteria including playing faster, removing silence gaps between spoken words, removing hesitation particles, and frequency shifting; and the ability to apply independent gain control to individual senders in a multi-party Conversation; as well as other potential Rendering options.

**[0114]** The Store and Stream module 24 manages control and informational messaging between itself and MCMS.

### D.2.1 The Persistent Infinite Message Buffer (PIMB)

**[0115]** The Persistent Infinite Message Buffer or PIMB 30 is a set of indexed (i.e., time-stamped and sequentially numbered) Media payload data structures and a system for their storage and retrieval. In one embodiment, the data in the PIMB 30 is arbitrarily persistent, meaning it is available virtually forever or at least until the system runs out of storage. Various retention rates and strategies may be employed to make effective use of storage resources. Many possible implementations exist for the physical storage implementation of the PIMB 30, including, but not limited to: RAM, Flash memory, hard drives, optical media, or some combination thereof. The PIMB 30 is also "infinite" in size, meaning the amount of data that can be stored in the PIMB 30 is not inherently limited. This lack of limit is in comparison to existing jitter buffer technology that

discards data as soon as it is rendered. In one specific embodiment, the PIMB 30 may be implemented using a small and relatively fast RAM cache memory coupled with a hard drive for persistent storage. As the physical storage capacity of the PIMB 30 is exceeded, the data is maintained on the Server 16 (as described below) for later retrieval on demand. User criteria or a replacement algorithm, such as least-recently-used, or first-in-last-out, is used to control the actual data stored in the PIMB 30 and the data that is maintained on the Server 16 or archived at any point in time. The PIMB 30 further provides the attributes of file system storage and the random access attributes of a database. Any number of Conversations, regardless of their duration or the number of Messages in each, may be stored and later retrieved for Review. In addition, the meta data associated with the Messages of a Conversation, such as its originator and its length, may be also stored in the PIMB 30. In alternative embodiments, the indexed Media payloads and other data can be stored for a designated period of time (e.g. 30 days). Once the age of the media exceeds the designated period, the payloads and data are discarded. In another embodiment, payloads can be discarded based on the sender and/or the recipient of the Message containing the payload, or the topic of the Conversation or Messages associated with the payload. In yet other embodiments, payloads and data may be marked for transience, meaning the Messages will not be stored in the PIMB 30 beyond the requirements for immediate Rendering.

#### D.2.2 The Data and Network Quality Store

**[0116]** The data and network quality store (DNQS) 32 is a data store for storing information regarding Media payloads and Vox packets that are either read from or written to the PIMB 30.

#### D.2.3 The PIMB Writer

**[0117]** The PIMB writer 28 writes data to the PIMB 30 for two basic purposes. The PIMB writer 28 writes data from a Media capturing device (e.g., a microphone or camera) on the Device 13 running the Client 12 ("Encode Receive"). The PIMB writer 28 also writes data received over the network 18 from other Clients 12 into the PIMB 30 ("Net Receive").

##### D.2.3.1 Encode Receive

**[0118]** For capturing Media from the Device 13, the PIMB writer 28 includes Encoder Receiver 28a and a Data Storer 28c. When a User speaks into the microphone or

generates video images with the Device 13 for example, the hardware 34 receives the raw audio and/or video signals and provides them to the Encoder Receiver 28a, which encodes the signals into indexed media payloads (hereafter sometimes simply referred to as "payload" or "payloads"). The Data Store 28c stores the payloads into the PIMB 30. Other types of Media, such as sensor data is converted into payloads in a similar manner.

#### D.2.3.2 Net Receive

**[0119]** For storing Media received over the network 18 into the PIMB 30, the Net Receive function of PIMB writer 28 includes a Network Receiver 28d, a Data Bufferer 28e, a Data Storer 28f, a Data Quality Manager 28g, and a Network Quality Manager 28h. The Network Receiver 28d receives Vox packets over the network 18. The Data Bufferer 28e places the received Vox packets into their proper sequence and prevents the Rendering of the incoming Vox packets for at least the Minimum Time Shift Delay (MTSD) delay. The Data Storer 28f transforms the packets into indexed media payloads and stores the indexed media payloads in the PIMB 30. As the payloads are stored, the Data Quality Manager (DQM) 28g notes any missing or defective packets. If packets are missing or defective, the DQM 28g schedules a request for retransmission over the network 18. The sending device replies by resending the missing or defective packets. Eventually these packets are converted into indexed media payloads and stored in the PIMB 30. By retrieving the missing or defective packets, an "exact" copy of a sender's Message is eventually stored in the PIMB 30. The retransmission of missing and/or defective packets does not delay the rendering of Messages in real time, provided that the packets that have been delivered are of "good enough" quality and quantity. Retransmission requests may be delayed by the DQM 28g if insufficient network resources are available to support both the new "live" data as well as the retransmission.

### D.2.4 The PIMB Reader

**[0120]** The PIMB reader 26 reads data from the PIMB 30 for two basic purposes. The PIMB reader 26 accesses the PIMB 30 when data is to be rendered ("Render") for the local Client 12. Data is also read from the PIMB 30 when data is to be transmitted ("Transmit") by the Client 12 over the network 18.

#### D.2.4.1 Render

**[0121]** For the rendering of messages on the Client 12, the PIMB reader 26 includes a Data Prioritizer 26a, a Data Retriever 26b, a Packet Loss Compensation/Interpolator ("PLC/Interpolator") 26c, a Data Mixer 26d and a Data Renderer 26e. The Prioritizer 26a prioritizes the data to be rendered by building an ordered queue of messages that could potentially be Rendered. It uses User configured Priority for the rendering of consecutive Conversations (MCMS-C). In addition, the Data Prioritizer uses the availability of media data to render within the limits imposed by the MTSD, the User's current Attention, and the User's defined and implied Intentions. The Data Retriever 26b retrieves the prioritized indexed media payloads from the PIMB 30. The PLC/Interpolator 26c performs packet loss compensation and interpolation on the retrieved payloads, using known packet loss compensation and interpolation algorithms. The particular method to be used is dependent on the media Codecs in use, and other well-known parameters. The Mixer 26d is used to appropriately mix concurrent data streams from multiple Messages of a single Conversation together. For example, if two or more Participants of a Conversation are speaking at the same time, the Mixer 26d mixes the Messages, creating the effect of both Participants speaking at the same time. In an alternative embodiment, the User has the option of Reviewing the multiple streams from one Participant at a time. If only one Participant in the Conversation is speaking, the Mixer 26d may simply pass the single Message stream, without performing any mixing. The Renderer 26e takes the data from the Mixer module 26d and converts it into a form suitable for the hardware driver 34. The hardware 34 then drives either the speaker or video display of the Device 13, depending on the type of Media, creating voice, video or some other audible and/or visible notifier on the Device 13.

#### D.2.4.2 Transmit

**[0122]** To prepare messages for transmission from the Client 12 over a network 18, the PIMB reader 26 includes a Data Prioritizer 26f, a Packet Quality Manager (PQM) 26g, a Data Retriever 26h, Packetizer 26i, a Transmitter 26j and an Acknowledger 26k. The Data Prioritizer 26f prioritizes the Messages for transmission over the network 18. Priority is determined using the MCMS Participant Attributes related to payloads available for transmission, network connectivity and bandwidth conditions, and the Intentions of Users beyond the next hop to either Render live or time-shifted, and, in

some embodiments, possible optimizations of transmission bundling where multiple packets to any given next network hop are available. The prioritized packets are then optimized, using the PQM 26g, which assures the timely delivery of "good enough" data quality for live Messages, while minimizing real-time bandwidth, as described in more detail below. The Data Retriever 26h retrieves the appropriate payloads from the PIMB 30. The Packetizer 26i assembles the payloads into Vox Packets, which are then transmitted by the Transmitter module 26j over the network 18. When the recipient receives the Vox packets, an acknowledgement is sent back to Acknowledger 26k over the network 18 for notifying the sending User that a Message has arrived at its destination.

### D.2.5 The Packet Quality Manager

**[0123]** The PQM 26g has several optimization goals: (i) the timely delivery of an adequate copy of time-sensitive Media ("as soon as possible to be good enough" for Rendering); (ii) the efficient use of available bandwidth, meaning using the optimal transmission frequency, payload quality, and packet size for the underlying network; and (iii) the ability to dynamically adjust or make changes in the transmission frequency, payload content, payload quality, packet size, etc. as network conditions change.

### D.2.6 The Network Quality Manager

**[0124]** On the receiving side of a network transmission is the Network Quality Manager 28h (NQM). The NQM is responsible for observing specific properties of network performance for each sender that has sent media to the Client 12, comparing expectations of jitter, loss, and throughput to their actual values. This is used to compute a Network Quality Rating (NQR) for every sender. This NQR is used to indicate sender availability and Conversation live-ness to the User of the receiving device.

### D.2.7 The Data Quality Manager

**[0125]** The Data Quality Manager 28g measures the quality of the data being received over the network by observing packet loss, jitter, and throughput. The DQM 28g uses these measurements for three purposes: (i) to send receipt reports back to the sender; (ii) optionally using those receipt reports to request retransmission of certain data; and (iii) making these measurements available to the NQM 28h.

## E. Server Architecture

**[0126]** Referring to Figure 3, a block diagram of the application 78 that runs on the Server(s) 16. The application 78 is similar to the Client application 12 in many regards and includes an MCMS server application 80, an MCMS database 82, a store and stream module 84, a PIMB 85, a data and network quality store (DNQS) 86, MCMS-SAS message handling modules 87a and 87b which manage messages and signals back and forth between the MCMS server application 80 and Store and Stream module 84, an archive/retriever 88, and an archive 89. The application 78 further includes an authentication-encryption-security module 40 and a communication protocol module 44.

**[0127]** The MCMS server application 80 is a multi-tiered architecture including a MCMS Database Module 20a, a Store and Stream (SAS) Module 20b, a Messaging/Signaling Module 20c, Conversations/Messages management services 20f, Priority services 20g, Contacts (including User and Authentication) services 20h, Presence/Status service 20i, and Messages/Signals services 20. The aforementioned modules and services of the application 78 are similar or the same as the modules and services having like reference numerals as the Client 12, and therefore are not described in detail herein, except for one notable exception. In various embodiments, the MCMS server application 80 and Store and Stream module 84, including the MCMS database 82, is configured to support many Users in one instance of the application. The one instance may be further configured to support multiple domains, where each domain is defined as a group of Users (i.e., a corporation or other group of Users belonging to a common organization). This architecture allows each application 78 on Server 16 to serve multiple users (or domains), where each user (or domain) is not visible to other Users. This partitioning is referred to as "multi-tenancy".

**[0128]** The server Store and Steam 84 module performs the functions of Net Receive and Transmit. For the Net Receive function, the module 84 includes a Net Receiver 28d, Data Bufferer 28e, a Data Storer 28f, a Data Quality Manager (DQM) 28g, and a Network Quality Manager 28h. For Transmit functions, the module 84 includes a Data Prioritizer 26f, Packet Optimizer 26g, Data Retriever 26h, Packetizer 26i, Transmitter 26j and an Acknowledger 26k. The aforementioned elements of the Store and Stream module 84 are similar or the same elements having like reference numerals as the Client 12, and therefore are not described in detail herein.

**[0129]** Since the Server 16 does not directly interact with Users, the encoding and rendering functions provided in the Store and Stream module 24 of the Clients 12 need not be present. The MCMS application 80, when running on Servers 16, does not interact directly with Users. Consequently, the user interface and user interface API modules and services 20e and 20d are not needed.

**[0130]** The application 78 on each Server 16 potentially serves multiple tenants, meaning it serves multiple Users of the system 10. The PIMB 85 of the server application 78 is therefore significantly larger and is used to store the media payloads of multiple Users, as opposed to the PIMB 30, which is used to store just the generated or received payloads of only a single User. The main purpose of the Store and Stream module 84 is to receive Messages transmitted by the Clients 12 and transmit Messages to other Clients 12. As Messages are received, they are stored in the PIMB 85 and transmitted to the next Server 16 (i.e., the net "hop") of the network layer 14 along the path to the intended recipient(s), or to the recipient(s) directly depending on the system configuration. The archive-retriever 88 is responsible for archiving the media payloads stored in the PIMB 85 in archive 89. As the physical space in the PIMB 85 is exhausted, media payloads in the PIMB 85 are moved to the archive 89, which is a mass storage device. In various embodiments, the payloads stored in the PIMB 85 may be archived in accordance with User defined criteria and/or any known replacement algorithm, such as first-in-first-out (FIFO) or least recently used (LRU). It should be noted that only a single Server 16 is illustrated in Figure 1 for simplicity. It should be understood that in actual embodiments, multiple servers or a "server farm" may be used for a network with a large number of Users.

**[0131]** The terms "persistent" and "infinite" used to describe the PIMB 30 and the PIMB 85 should not be construed literally as absolute terms. A User may wish to indefinitely store some Messages that are considered important. In other situations, such as a casual chat between two friends, the Messages may be deleted after a certain period of time to save space. According to various embodiments of the invention, different retention policies may be used, either set by the system 10 or configured by the User. The use of the word "infinite" refers to the lack of any preset time boundaries enforced by the PIMB. This is contrasted with current jitter buffer systems, which discard media after it has been rendered. The terms persistent and infinite should therefore be broadly

construed to mean that the PIMB 30 and PIMB 85 have no internal limitations on the time ranges and quantities of Messages that can be stored therein.

**[0132]** There are a number of advantages to archiving the Messages of Conversations in a persistent storage medium. Voice messages and other Media can be organized, indexed, searched, transcribed, translated, and Reviewed as needed. Voice, as well as other Media, therefore becomes an asset that can be managed both by Users and organizations. These Media assets have value to corporations, first responders, police and fire departments, as well as the military.

## F. The Vox Protocol and Indexed Media Payloads

**[0133]** As noted above, the Vox protocol is used by the Store and Stream module 24 to support all facets of payload transmission, storage, and optimization. The Vox packet is a structured message format designed for encapsulation inside a transport packet or transport packets of the underlying technology of the network 18. This arrangement significantly improves the flexibility of the system 10. By embedding the Vox packets into existing transport packets, as opposed to defining a new transport layer for "Voxing" applications, the system 10 takes advantage of current packet based communication networks running over the existing telecommunications infrastructure. A new network infrastructure for handling the Vox packets therefore need not be created to take advantage of all the benefits of the system and method described herein.

**[0134]** Referring to Figure 4A, the general format structure of a Vox packet 95 is illustrated. The format of the Vox packet 95 includes fields for type, sub-type, length, and payload. The different types of Vox packets include authentication, signaling, media payload, media multiplex (one message), and media multiplex (multiple messages). The sub-type field is used to designate different types of authentication, signaling or media type messages. Possible sub-types for authentication Messages include those necessary for key exchanges and Authentication. Possible sub-types for signaling Messages include registration, routing, message set-up, and network management. Possible sub-types for Media messages include different Codec styles and different payload aggregation techniques. The length field defines the overall length or size of the payload. The payload field contains the actual payload or media of the packet 95.

**[0135]** Referring to Figure 4B, a diagram illustrating a Vox packet 95 encapsulated in an exemplary protocol used by the network 18 is shown. In this example, the Vox packet 95 is embedded in underlying UDP, IP and Ethernet transport packets 96 respectively. In this manner, the Vox packet 95 can be transported across underlying UDP, IP and Ethernet layers of the network 18. This is a standard protocol encapsulation technique used by packet networks.

**[0136]** Referring to Figure 4C, a diagram illustrating a media multiplex Vox packet 95 encapsulated in UDP, IP, and Ethernet 97 is illustrated. In this example, the Vox packet 95 includes a Media type field, a Media sub-type field, a length field, a message ID field, a time stamp field, a sequence ID field, and a Media payload field.

**[0137]** Referring to Figure 4D, the format of an indexed media payload 98 is illustrated. The indexed media payload includes a sub-type field, a length field, a message identifier (ID) field, a time-stamp field, a sequence identifier (ID) field, and field for the Media payload.

**[0138]** The encapsulation of Vox packets 95 into the transport packets of the underlying network allows the Media, Messages and Conversations to each be defined by a number of attributes.

**[0139]** When Media is created or otherwise originated on a Device 13, it is typically time-based, meaning it changes in some meaningful way over time. As a person engages in a Conversation for example, their spoken words are strung together into sentences or statements, which vary over time, and the resulting data (streams and packets) will maintain the same variance over time. Similarly video (as opposed to a still photo) as well as GPS or other sensor data will vary over time. Regardless of the type or how it is originated, the Media is segmented and placed into the payloads of a plurality of Vox packets 95. The packets are then continually, stored, transmitted, received, stored and rendered in streams (i.e., streaming media) at the transmitting and receiving Devices 13 respectively. Since each packet 95 is indexed, time-stamped, and given a sequence identifier, the individual packets can be segmented into Messages. By sequentially threading the individual Messages together, Conversations may be constructed.

**[0140]** One further unique aspect of the system 10 is that the media payloads generated by a Client 12 are stored in multiple locations. Not only are the payloads stored in the PIMB 30 of the generating Device 13, but also in the PIMB 85 of the Server(s) 16 and

the PIMB 30 of the receiving Devices 13. This basic feature enables or makes possible much of the Voxing functionality described above and provides the system 10 with both resilience and operability, even when network conditions are poor or when a Participant of a Conversation is not connected to the network.

### G. Interoperability With Underlying Telecommunication Protocols

**[0141]** The system 10 is intended to run or be layered over a variety of existing communication networks 18, such as the Internet, fixed PSTN type circuit networks, and mobile or cellular phone networks, or a combination thereof. The system 10 is designed around the concept of moving many small units of information (i.e., the Vox packets) between different Clients 12 and Servers 16 in the system 10. While Vox packets may vary in size, depending on their function and payload, they all appear to be the same kind of data to the underlying network layer. In one embodiment, the system 10 has been designed and optimized for IPv4 networks such as the Internet, but other types of networks may be supported as well. For the purposes of this document, the term "IP" should be taken to mean IPv4, IPv6 or any other current or future implementation of the Internet Protocol.

**[0142]** Referring to Figure 5, a diagram of a Client 12 running on Device 13 and communicating with a Server 16 over a shared IP network 100 is shown. As illustrated, the Client 12 is coupled to the shared IP network 100 through a first Internet service provider A and the Server 16 is coupled to the shared IP network 100 by a second Internet service provider B. During communication, the Vox packets 95 (designed "VP" in the figure) are encapsulated within UDP/IP packets and then interleaved among other IP protocol packets as is well known in the art and transmitted across the shared IP network 100 from the Client 12 to Server 16, or vice versa. As is well known, each lower packet layer encapsulates the entire packet of the layer immediately above it. Packets can also be sent in a similar manner between two Servers 16. Messages are sent from one Client 12 enabled Device 13 to another over a shared IP network 100. At each hop, the Vox packets 95 are embedded in the underlying IP protocol and transmitted, until they reach the target destination.

**[0143]** The diagram of Figure 5 is merely exemplary, showing only a single Client 12 and Server 16 connected to the network 100 for the sake of illustration. In actual embodiments of the system 10, a large number of Clients 12 and one or more Servers 16

are typically connected to the shared IP network 100. It is also useful to note that the Client 12 and Server 16 do not have exclusive use of the IP network 100. In the example shown, an HTTP client, which is coupled to the network 100 through Internet provider A, can send packets back and forth with an HTTP server, coupled to the network 100 through a third Internet provider C. The system 10 does not control the manner in which the VPs embedded in the IP packets traverse the network 100. Rather, all packets that traverse and share the network 100 do so in accordance with the standard procedures of the underlying shared IP network 100.

**[0144]** Referring to Figure 6, a "circuit" based network 104 such as a GSM mobile phone network is illustrated. The circuit network 104 is coupled between Client 12 running on Device 13 and Server 16. Once a circuit is established between the Client 12 and Server 16, the system 10 layers Vox packets (VP1, VP2, VP3, VP4, VP5, etc.) onto the underlying packets used by the network 104 and transmits them across the network 104, creating "virtual Vox" circuit. The Vox Packets sequentially traverse the circuit network 104, typically with spacing or framing data as is well known in the art for transmitting data over a circuit network. In addition, packet construction parameters, such as the payload size and the number of header fields included may be used to exploit the lack of per-packet overhead and to increase speed and/or efficiency of data transfer across the network 104. It should be noted again that for the sake of simplicity, only a single Client 12 and Server 16 are shown connected to the network 104. It should be understood, however, that additional circuits between Clients 12 and Servers 16 as well as other components may be established concurrently through the network 104. The network 104 is therefore not dedicated for the transmission of Vox Packets, but rather may be shared with other types of network traffic.

**[0145]** Referring to Figure 7, a diagram illustrating communication between a first Client 12A enabled Device 13A associated with a first network A and a second Client 12B enabled Device 13B associated with a second network B is illustrated. The networks A and B further each include gateway Servers 16A and 16B respectively. The Gateway Server pair 16A and 16B facilitate communication between the two networks A and B, allowing the Devices 13A and 13B to communicate with each other. In various embodiments, the networks A and B could each be any type of network. For example, each network A and/or B could be an IP network, a circuit type network, or a wireless or cellular network (i.e., CDMA, GSM, TDMA, etc.). The Servers 16 that straddle the two

networks A and B are considered gateway servers because they route traffic or serve as a "gate" between the two networks.

**[0146]** With the system 10, there are a several basic network interaction considerations to optimize system performance. These considerations include factors such as resolving the underlying address to which the Vox packets 95 are to be sent, the integrity of any sent Vox packets, and the management of the Maximum Transmission Unit (MTU) of a single Message that may be sent across a given network or combination of networks.

**[0147]** The address of a target Client 12 needs to be known so that the underlying network delivers the Vox packet 95 to the correct location. With IPv4 networks, the address is typically an IPv4 Address, which is a 32-bit number that uniquely identifies a host within the network. For other networking technologies, the address could be some other type of identifier. IP networks use the Domain Name System (DNS) to resolve human-readable names into IP addresses, and the Address Resolution Protocol (ARP) to resolve IP addresses into physical addresses. Regardless of the underlying networking technology, the system 10 uses one of the above-mentioned or other known addressing schemes for delivery of Vox packets 95 to the correct location.

**[0148]** As with almost any packet-based communication system, transmitted Vox packets might not be delivered to their addressed location if the underlying network is unable to deliver the packets in which the Vox packets are encapsulated. Most packet networks do not inform transmitters when packets are dropped. Instead they rely on the transmitters and receivers to notice and compensate for any dropped packets. The system 10 is designed to use these receiver receipt report messages to coordinate this packet loss management. If the underlying network is able to inform the sender of lost or dropped packets, the system 10 utilizes this information in its retransmission protocol.

**[0149]** The management of MTU is the determination of the Maximum Transmission Unit (i.e., the maximum size of a single message) that may be sent across a network. For packet-based networks, the underlying network imposes the MTU. For circuit-switched networks, the MTU may be a tunable parameter for network efficiency and performance. Thus in most cases, the underlying network imposes or determines the maximum size of the Vox packet 95 that may be transmitted efficiently. For example with IP networks, packets may be fragmented if the payload exceeds the MTU, but at a

substantial performance penalty. With IP over Ethernet networks, the transmitting device has an MTU of 1518 bytes, as enforced by Ethernet. The largest IP packet must leave room for the Ethernet headers. The largest UDP packet must leave room for both IP and Ethernet headers and the largest Vox protocol that may be generated on Ethernet for example is the Ethernet MTU (1518) – IP header (20) – UDP header (8) = 1490 bytes. Since the Vox protocol will have a header of its own, the actual Vox media payload will be less than 1490 bytes on an Ethernet network. For Gigabit Ethernet, the MTU could be much larger, but would be determined using a similar formula.

**[0150]** In a purely packet-based network, there are two potential values for MTU, the local link MTU and the path MTU. Determining the local link MTU yields the maximum size for Vox packets to be efficiently sent out to the local network interface. The path MTU yields the maximum size of Vox packet that may be sent intact all the way to the remote node. If a sender is connected via Ethernet, the Vox packet might pass through various other systems with smaller MTUs en-route to the recipient. The smallest MTU on the path to the destination needs to be resolved and known by the sender. In the IP world, there is a standard procedure for discovering the smallest MTU called "Path MTU Discovery". For other kinds of networks, an equivalent procedure may be used. Again, since the system 10 is layered on top of other networks, any of the above MTU algorithms may be used.

## H. Operation Flow Diagrams

### H.1 Store and Stream

**[0151]** Referring to Figures 8A through 8F, a series of flow diagrams are provided to illustrate the operation of the store and stream module 24 and 84 on Clients 12 and Servers 16 respectively. Figure 8A shows the sequence of operation for a first Client $12_1$ transmitting Messages to a second Client $12_2$. Figures 8B and 8C illustrate the operation of the PIMB writer 28 and PIMB Reader 28 on the transmitting Client $12_1$. Figures 8D and 8E illustrate the operation of the PIMB Writer 28 and PIMB Reader 26 on the receiving Client $12_2$. Figure 10F illustrates a flow diagram of the Store and Steam module 84 on a server 16.

**[0152]** In Figure 8A, the User of Client $12_1$ running on a Device $13_1$ originates Media to be transmitted. The Media can be originated at the Device 13 in a number of different ways such the User creating Media by speaking into a microphone or creating video

content on their Device 13. Media is also originated by a Device 13 by receiving sensor data, such a GPS information or a temperature reading. Regardless of how the Media is originated, the Media is encoded by the PIMB Writer 28 (box 130), which converts the Media into indexed media payloads and stores them in the PIMB 30 (box 132) on Client $12_1$. The PIMB Reader 26 on the Client $12_1$ reads the payloads out of the PIMB 30, creates Vox packets, and transmits the packets to the receiving Client $12_2$ (box 134) over the network 18. Each Server 16 along the path between the sending Client $12_1$ and the receiving Client $12_2$ stores the transmitted payloads in the PIMB 85 and transmits the Vox packets to the next hop (box 133). At the receiving Client $12_2$, the net receive function of the PIMB Writer 28 converts the Vox packets into indexed media payloads (box 136) and stores the payloads into the PIMB 30 of Client $12_2$ (box 138). The rendering module of the PIMB reader 26 on Client $12_2$ renders the payload information read from the PIMB 30 into a medium suitable for human consumption, such as voice or video (box 140). Each of these steps are described in more detail below with respect to Figures 10B through 10E.

**[0153]** In Figure 8B, a sequence of the Encode Receive function performed by the PIMB Writer 28 (step 130 of Figure 8A) is provided in detail. In the initial step $130_1$, the User of the Device 13 running the Client $12_1$ originates the Media to be transmitted. As noted above, the Media may be derived by speaking into a microphone, using a camera, receiving sensor data, or by some other Media generating component. In the next step $130_2$, the Encode Receiver 28a encodes the Media and creates the indexed media payloads (step $130_3$), which are then stored in the PIMB 30 (step 132) by the Data storer 28c.

**[0154]** In Figure 8C, the sequence of the Transmit function performed by the PIMB Reader 26 (step 134 of Figure 8A) on the sending client $12_1$ is provided in detail. In decision loop $134_1$, the transmit function of the PIMB Reader 26 continuously checks to see if indexed media payloads that are to be transmitted have been written into the PIMB 30 and are available for transmission. If such payloads are available in the PIMB 30, the Data Prioritizer 26f prioritizes the payloads that should be sent first, using the MCMS Participant Attributes information, as illustrated in step $134_2$. Information about the highest priority payloads are passed to the Packet Optimizer module 26g which runs the PQM (step $134_3$), as described in more detail below with respect to Figures 9A-9C. The appropriate payloads are then retrieved from the PIMB 30 (step $134_4$) by the Data

Retriever 26h and converted into Vox packets 95 by the Packetizer 26i (step $134_5$). The Vox packets 95 are then transmitted (step $134_6$) by the Transmitter 26j over the network 18 to the receive Client $12_2$, which sends back receipt reports reflecting the properties (loss, jitter, throughput) of the packets that have been received. These receipt reports provide the information necessary for the PQM to calculate the MABR for a given receiver. The aforementioned process is repeated for each transmission loop as indicated by the return arrow from the transmit step to the top of the flow chart.

**[0155]** In the embodiment described above, the media is encoded, stored in the PIMB 30 and then transmitted over the network in a serial fashion. In an alternative embodiment, the encoded media can be stored in the PIMB 30 and transmitted over the network in parallel, meaning the two functions occur substantially at the same time.

**[0156]** In Figure 8D, the sequence for the Net Receive function (step 136 of Figure 8A) of the PIMB Writer 28 on the receiving Client $12_2$ is illustrated. In the initial step $136_1$, the network receiver 28d receives the Vox packets 95 over the network 18. The Data Storer 28f converts the packets into indexed media payloads (step $136_3$), which are stored in the PIMB 30 (step $136_4$). As the payloads are stored, the Data Quality Manager (DQM) 28g is run. The DQM 28g checks for missing or corrupted packets, ensures the eventually storage of an exact copy of the transmitted data, and the sends receipt reports regarding the conditions of the network to the transmitter. Each of these functions of the DQM 28g are described in more detail below with regard to Figures 9D through 9F.

**[0157]** In Figure 8E, the sequence for the Render function of the PIMB Reader 26 (box 140 of Figure 8A) on the receive Client $12_2$ is illustrated. In the initial step $140_1$, the Data Prioritizer 26a prioritizes the indexed media payloads to be rendered as determined by the MCMS application 20 using the MTSD information as well as User status and presence information, including the User's Intentions and Attention status. The prioritized payloads are then read from the PIMB 30 (step $140_2$) by the Data Retriever 26b. The PLC/Interpolator 26c performs packet loss compensation and interpolation (step $140_3$) on the retrieved payloads, using known packet loss compensation and interpolation algorithms depending on which Codecs are used. In the next step, the Mixer 26d mixes (step $140_4$) multiple Messages of a Conversation if two or more Participants have generated Media at the same time within the same Conversation (e.g., both are speaking at the same time). The Renderer 26e renders (step $140_5$) the data

stream from the Mixer 26d, generating sound, video, or other Media (step $140_6$) for the recipient User.

**[0158]** In Figure 8F, the sequence for a Server 16 to receive Vox packets from the previous hop on the network 18, store, archive and transmit the Vox packets to the next hop is illustrated. In the initial step, the Server 16 performs the Net Receive function of the PIMB writer (similar to Fig. 8D) to store the indexed media payloads of the received data in the PIMB 85 and archive 89 or the Server 16. The server 16 also performs the Transmit function of the PIMB writer (similar to Fig. 8C) to forward the received packets onto the next hop on the network 18. In this manner, a copy of the media generated by the transmit Client $12_1$ is received, stored and transmitted at each hop along the path to the receive Client $12_2$.

**[0159]** In the aforementioned embodiment, the writing of received indexed media is stored in the PIMB 91 of the Server 16 and transmitted to the next hop in a serial fashion. In an alternative embodiment, the received indexed media payloads can be stored in the PIMB 91 and transmitted to the next hop substantially at the same time. The storage of Media on the PIMB 30 of both transmitting and receiving Devices 13 allows for the progressive transmission and rendering of Media. On the transmit side, the Media originating on the transmitting device may be progressively transmitted over the network as it is being received. In various embodiments, the encoded Media (regardless of how it is originated) may be progressively transmitted before, after, or at substantially the same time it is stored in the PIMB 30. On the receive side, the incoming Media may also be progressively rendered as it is received over the network, provided the User has opted to Review the Media in the near real-time mode. In various embodiments, the incoming Media may be progressively rendered before, after or substantially at the same time as it is stored in the PIMB 30 of the receiving Device 13. If the received Media is to be Reviewed in the time-shifted mode, then the Media is retrieved from the PIMB 30 (or possibly a PIMB 85 on a Server 16 if replaced on the local PIMB 30) for later Review at a time designated by the User.

**[0160]** In the context of the present application, the term progressive or progressively is intended to be broadly construed and generally mean the continuous processing of a data stream based on availability of the data. For example, as Media is created or otherwise originated on a Device 13, the progressive encoding, storage, packetization and transmitting of that media is continuous, so long as the Media is available. As a

person speaks, that Media is progressive or continuously encoded, stored, packetized and transmitted for the duration of the persons speech. When the person pauses or stops speaking, there is no media to progressively process. When the person resumes speaking again, the progressive processing of the Media resumes. On the receive side, the Media is also progressive processed as the Media is being received (i.e., available). As Media the Media is received it is continuously stored. It will also be continually rendered as the Media is being received when in the near real-time mode or from storage when in the time-shifted mode. Although the above explanation was provided in the context of voice, it should be understood that all types of Media can be progressively processed in a similar manner. Also the progressive processing of Media does not necessarily have to be progressively processed in time-indexed order. Rather the Media is processed in the order in which it is received. If Media is received out of index order, in one embodiment, the Media is progressively processed in the order it was received and then organized into the indexed sequence in the PIMB 30. In an alternative embodiment, the received Media can be organized into its indexed sequence and then progressively rendered.

### H.2 PQM Operation Flow Diagrams

**[0161]** The PQM 26g relies on a metric called the Maximum Available Bit Rate (MABR), which is a continually computed approximation of actual transmission capacity or bandwidth (i.e., a measure of the capability of the network at a given point in time) between a sending and receiving node pair. As instantaneous network conditions change, the MABR is updated. Regular measurements of network throughput, packet loss, and jitter are considered in computing the MABR. In an alternative embodiment, the MABR may also be manually set or limited based on time of day, type of network, other conditions or parameters.

**[0162]** The PQM also considers the Intention of the recipient(s) to optimize transmission for time-sensitivity. A transmission is considered time-sensitive if either (i) the Intention of the recipient(s) is to Review the transmission "live" or in the near real-time mode, or (ii) the recipient would like to immediately Review a Message that for some reason is not currently stored on their Device 13 (e.g., the Message was previously stored in the archive 89). The Intention of the recipient can be either inferred by the behavior of the recipient or the recipient may set or otherwise designate their Intention. On the other hand, a transmission is considered to be not time-sensitive if the

Intention of the recipient is to Review the Message in the time-shifted mode. The Intention of the recipient to Review the transmission in either the live or time-shifted mode at least partially defines the "timeliness requirements" of the transmission. In various other embodiments, factors such as the urgency or priority of transmissions may also be considered in defining the timeliness requirement of the transmission.

**[0163]** The nodes in the network path between a sender and a receiving pair also need to be consistent regarding the status of intentions of the recipients. If one target recipient indicates timeliness, meaning they wish to Review the transmission immediately or live, then all the intermediate nodes on the network along the sender-receiver path need to have the same timeliness requirement, regardless of the requirements of other recipients. The timeliness requirement of each of the intermediate nodes is therefore dependent on the receiving nodes the transmission is being sent to. This dependency is sometimes referred to as a "union of requirements" for target nodes in the network transmission path.

**[0164]** The PQM further considers an Ideal Bit Rate or "IBR" for each scheduled Message payload transmission. For time-sensitive transmissions, the IBR is computed based on the packetization rate needed for substantially real time or live communication (referred to herein as the Real Time Bit Rate or RTBR). With voice for example, a packetization rate of a packet every 20 milliseconds containing 20 milliseconds of audio data is considered an acceptable IBR for conducting live conversations. The RTBR for such a system in kilobits per second would be the size of 1 second of audio payload data plus the size of all network headers that would be generated for the transmission. For video media or a combination of voice and video, the RTBR will likely be substantially higher than simply voice. For other types of media such as sensor or GPS positioning data, the RTBR will likely be lower than that of voice. For non time-sensitive transmissions, the IBR is set to a Maximum Efficiency Bit Rate (MEBR) to optimize the usage or efficiency of transmissions over the network. The MEBR is calculated by adjusting the packetization rate to its largest possible value for the underlying network. If multiple messages or payloads are to be sent between a sending and receiving pair, then an Aggregate IBR (AIBR) is considered for the transmission.

**[0165]** The PQM operates by sending data in a series of transmission loops for each sending and receiving pair. The transmission loops for each sending and receiving pair are independent. Any transmission on the network may affect the MABR of other

sending-receiving pairs. Accordingly, the MABR is preferably continually computed for all recipients.

**[0166]** Referring to Figures 9A through 9C, flow charts illustrating the sequence of operation of the PQM for a single sending and receiving pair is illustrated. In Figure 9A, the steps in determining the MABR between the single sending and receiving pair are illustrated. In Figure 9B, a flow chart illustrating the steps for calculating the AIBR for each transmission loop for the single sending and receiving pair are illustrated. In Figure 9C, a sequence for determining the amount of data to transmit between the sending and receiving pair per loop is illustrated. The processes illustrated in the three diagrams run simultaneously and interact with one another, as described in more detail below.

**[0167]** In Figure 9A, a flow chart 50 for calculating the MABR for the network interface between the sending and receiving pair is shown. In the initial step $50_1$, the network interface between the sending and receiving pair is monitored. The sender periodically receives reports, which contain information regarding the status of the network connection at the receiver in step $50_2$. The reports include information regarding the current status of data throughput $50_3$, packet loss $50_4$, and jitter $50_5$ as observed by the receiver at the network interface. In step $50_6$, the MABR is calculated at the sender based on these observations contained in the reports. By monitoring or observing the data in these reports, the MABR value is continually adjusted based on current network capabilities or conditions between the sending and receiving pair. As network capabilities become more favorable for data transmission, the MABR will increase. If the network capabilities become less favorable for transmission, the MABR will decrease, potentially all the way to zero for an unusable network. The reports are similar to the packet loss reports generated by nodes in TCP networks, but additionally include throughput and jitter information as well.

**[0168]** If there are several network interfaces between the sending and receiving pair, an MABR is calculated for each interface for which a receipt report is received. If no traffic has been recently sent on the network, or no receipt reports have been received, the MABR may not reflect current network conditions. However, since receipt reports are generated continually by receivers while data is transmitted, the sender's MABR metrics will quickly converge to a more accurate value.

**[0169]** Referring to Figure 9B, a flow chart 52 illustrating the steps for calculating the AIBR for a transmission loop is illustrated. In the initial step $52_1$, the Messages with Media (by which we mean portions of the time indexed media that belongs to this Message) ready to be transmitted between the sending and receiving pair in the current loop are ascertained. A list of Messages with Media is then built $52_2$. For each Message in the list $52_3$, the time-sensitivity or timeliness requirement of each Message is considered $52_4$. If a particular Message is not time-sensitive, then the IBR is set to the Maximum Efficiency Bit Rate (MEBR) $52_5$. On the other hand, if a Message is time-sensitive, then the IBR is set to the Real Time Bit Rate (RTBR) $52_6$. In the next step $52_7$, the previously computed IBRs for each of the Messages in the list are summed together, resulting in the Aggregate Ideal Bit Rate (AIBR) $52_8$ for the transmission loop. As signified by the return arrow $52_9$, the above-described process is repeated for each transmission loop between the sending and receiving pair.

**[0170]** Referring to Figure 9C, a flow chart 54 illustrating the sequence for determining the rate of data to transmit between the sending and receiving pair per transmission loop is illustrated. In the initial step $54_1$, the MABR (as calculated in Figure 9A) is compared to the AIBR (as determined in Figure 9B) for the next transmission.

**[0171]** If the MABR is greater than or equal to the AIBR, then all the Messages identified as ready for transmission in the loop are packetized at the IBR rate $54_2$ and transmitted $54_3$.

**[0172]** On the other hand, if the MABR is less than the AIBR, then a series of procedures are applied to so that the PQM meets its goals of the timely delivery of an adequate copy of the data, the efficient use of available bandwidth, and/or adjustments to the payload content and quality, packet size, and transmission rate to meet current network conditions.

**[0173]** In the initial step, the Messages in the list are reviewed for time sensitivity $54_4$. If there are no time sensitive Messages, then the bit rate is reduced to the MABR $54_5$, and the Messages are transmitted $54_3$.

**[0174]** If the list includes time-sensitive Messages, the bit rate allocated for the non time-sensitive Messages is reduced $54_6$ until it meets the MABR limits. If reducing the bit rate all the way to zero is insufficient to meet the MABR, then these non time-sensitive Messages are removed from the list of Messages to be transmitted in the loop.

If the bit rate has been reduced so that it is less than or equal to the MABR, then the remaining Messages are Packetized and transmitted $54_3$.

**[0175]** If the removal of non time-sensitive Messages was not sufficient to meet MABR, then another procedure involving the use of a lower quality Codec (or Codecs) for the remaining time-sensitive Messages $54_7$ is used. An attempt is made to transmit the payload data as fast as possible by sending fewer bits during the transmission loop. In other words, by reducing the quality of the payload, the transmission sends fewer bits in a given period of time. In various embodiments, different Codec or Codecs, each having a different bit rate versus quality tradeoff, may be used. If the use of the lower quality Codec or Codecs is sufficient, meaning the MABR limit is met, then the Messages are transmitted $54_3$.

**[0176]** If the use of lower quality Codec(s) still does not meet the MABR, then the packetization interval of the time-sensitive Messages is increased $54_8$. With this procedure, the header-to-payload ratio is increased, which lowers the overall bit rate but introduces latency (i.e., a delay in the delivery of the transmission to the recipient). If this procedure results in the reduction of the AIBR to less than or equal to the MABR, then the transmission $54_3$ occurs.

**[0177]** If after changing the packetization interval the MABR is still not met, then the bit rate may be progressively lowered $54_9$ to be within the MABR limit. When the bit rate is lowered in this manner, time-sensitive messages are sent at a rate that is not sufficient to maintain a live conversation. Therefore, the Conversation is forced out of "live". If the network is down or conditions are very poor, it is possible that no data transmission may occur. Again, the aforementioned sequence is repeated for each transmission loop $54_{10}$ between the sending and receiving pair.

**[0178]** If there are multiple network interfaces between the sending and receiving pair, the sequence described in relation to Figure 9C is performed for each interface for which receipt reports are available. In various embodiments, the sender may contain logic to distribute the transmission load among the multiple interfaces. In different examples, the payloads can be sent only on one interface, while in other embodiments, some or all of the interfaces may be used.

**[0179]** The aforementioned description pertains to any sending and receiving pair in the system 10. In most situations, the sending and receiving pair will include a Client 12,

enabled Device 13 and Server 16, two Servers 16, a Server 16 and Client 12 enabled Device 13, or even possibly two Clients 12 respectively. If a sending node is transmitting to two (or more) receiving nodes at the same time, the above mentioned sequence as described in relation to Figures 9A-9C occurs concurrently for each receiving-sending pair.

### H.3 DQM Operation Flow Diagrams

**[0180]** The DQM 28g determines if data received at the Client 12 is either corrupted or if there are missing packets. In addition, the DQM 28g of a receiving Client 12 generates of the receipt reports, which are sent back to the transmitting node on the network. The DQM 28g also runs a background process to ensure that an exact copy of transmitted data is eventually received and stored. These functions are described below in Figures 9D through 9F respectively.

**[0181]** Referring to Figure 9D, a flow diagram illustrating the operation of the DQM 28g for checking for missing and/or corrupted data is illustrated. In the initial step $56_1$, the DQM 28g checks for corrupted packets using well-known techniques, such as CRC or similar integrity checking mechanisms. If a packet is corrupted, that packet is treated as missing $56_2$. The DQM 28g next ascertains if any packets are missing $56_3$. If an out of sequence packet is not received after a predetermined period of time, it is assumed to be missing. The DQM 28g notes any missing or corrupted packets $56_4$ in the DNQS 32. On the other hand if no corrupted or missing packets are detected, the DQM 28g determines if the quality of the received data was intentionally degraded by the sender $56_5$ for the purpose of saving bandwidth. The degraded quality is noted in the DNQS 32 $56_6$. Regardless if the quality of the received data is degraded or not, receipt information (e.g., a packet sequence number, time stamp, and the network address of the next node in the network the packet is to be sent) of the data is stored $56_7$ in the DNQS 32. The aforementioned process is continually repeated, as signified by the return arrow to the start of the flow diagram.

**[0182]** As a result of the process detailed in Figure 9D, information regarding the receipt of non-degraded packets, the deficiency of degraded quality packets, and missing packets, are all stored in the DNQS 32. As Media is received, the DNQS 32 maintains up-to-date information regarding the status of the Media.

**[0183]** Referring to Figure 9E, a flow diagram illustrating the operation of the receipt report generator function of the DQM 28g is illustrated. In the initial step, the DNQS 32 is periodically scanned $58_1$ to determine if there is any Media for which a receipt report needs to be generated $58_2$. If the answer is no, then the above scanning process is repeated. On the other hand if Media is identified, then the process determines if the Media is time-sensitive $58_3$, meaning either the User intends to Review the Media live or the user would like to immediately Review Media that is not stored on their Device 13.

**[0184]** If not time-sensitive, then the recipient informs the sender to set the retransmission priority (as defined below) to low $58_4$. If time-sensitive, then the amount of packet loss is considered $58_5$. If the amount of packet loss is outside a usable quality range, the retransmission priority is set to high $58_6$. As noted above, if the amount of packet loss is too large, the recipient may not be enabled to Review the Media upon receipt. If the quality is within an acceptable range, meaning the quality of the transmission is sufficient that it can be understood when rendered, then the priority for the sending of the receipt report is set to low $58_4$. Regardless if the recipient is Reviewing upon receipt or not, a receipt report is sent $58_7$, the DNQS 32 is updated $58_8$ and the Network Quality Manager (NQM) 28h is updated $58_9$. The retransmission requests defined in step $58_4$ is therefore conditional based on time-sensitivity. The transmission request defined in step $58_6$ is conditional on both time-sensitivity and quality.

**[0185]** The retransmission priority informs the PQM 26g of the sender to properly prioritize the transmission rate for the Media that requires retransmission. For example when the retransmission priority is set to high, then the sending PQM 26g of the sender should send any retransmissions before any new Media. If the priority is low, the PQM 26g should send the retransmitted Media after any new Media.

**[0186]** The aforementioned process is continuously repeated so that receipt reports are generated as Media is received. If the sender does not receive receipt reports in a timely manner, the PQM 26g of the sender will reduce the transmission rate, eventually stopping the transmission if no receipt reports are received.

**[0187]** Referring to Figure 9F, a flow diagram illustrating the sequence for requesting of missing or degraded Media is illustrated. In the initial step $60_1$, the DNQS 32 is

periodically scanned for missing or degraded Media $60_2$. If there is no missing or degraded Media, then the above defined scan is periodically repeated.

**[0188]** Media is considered missing if an out of sequence packet does not arrive after a predetermined threshold period of time $60_3$. If the packet arrives before the threshold, then it is no longer considered missing. On the other hand if the packet does not arrive after the threshold is exceed, then it is deemed missing. With missing packets, a low priority request for retransmission is made $60_4$ and the time of the request is noted $60_5$ in the DNQS 32. This process is repeated until the missing packet is received. When the missing packet arrives and the corresponding Media is available in the PIMB 30, the missing status of the Media is removed from the DNQS 32. The retransmission request defined in step $60_4$ is therefore conditional based on whether the Media is determined to be missing.

**[0189]** If degraded, the DQM 32 determines if the Media is part of a live Conversation $60_6$. If not, a request for a full quality copy of the degraded Media is made $60_7$, the full quality Media is designated as missing $60_8$ and the request time is noted $60_9$, in the DNQS 32. If the Media is part of a live Conversation, then no action is immediately taken in order to preserve network bandwidth. When the Conversation transitions out of the live mode, then the steps $60_7$ through $60_9$ are performed to ensure that a full quality (i.e. an exact or perfect) copy of the degraded Media is eventually received. When the data becomes available in the PIMB 30 of the recipient Client 12, the degraded status of the associated Media is removed from the DQNS 32. The transmission request defined in step $60_7$ is conditional on whether the Media is both degraded and not part of a live conversation.

**[0190]** The aforementioned process is continually repeated, as signified by the return arrows from $60_5$ and $60_9$ to the top of the flow diagram at $60_1$. By repeating the process outlined in Figure 9F, exact copies of all transmitted Media is eventually stored in the PIMB 30 of the receiving Device 13. In this manner, the storage of exact copies of transmitted Media is guaranteed at the recipient Device 13.

## I. Graphical User Interface

**[0191]** Referring to Figure 10, an exemplary Device 13 running Client application 12 is illustrated. The Device 13 includes a graphical user interface display 110, data entry buttons, keys, or keyboard 112, a microphone 114, and a transducer 116 for converting

electrical signals into sound, such as a speaker. The display 110 may also accept input as a touch screen. Further, the display 110 and keyboard 112 may be combined using a touch screen interface. As noted above, the Device 13 may be a variety of different communication tools, such as a desktop computer, laptop or other mobile computers, personal digital assistants, a programmable land-line or cellular phone, or programmable radios, or just about any other type of programmable communication device. The exemplary Device 13 illustrated in the figure is intended to be "generic", in the sense that it is supposed to represent or encompass all of the communication devices listed above. In addition, the term "graphical" user interface should not be construed as limiting. Other types of user interfaces which can be implemented on a Device 13 as well including Audio/DTMF interfaces, Voice User Interfaces (VUI), radio switch interfaces, or a combination thereof, can all be used to implement the various functions described below. For the sake of simplicity, each of these types of methods by which a User may interact with their Device 13 are generically referred to as a "user interface".

**[0192]** All Devices 13, regardless of their type, preferably have a user interface that enables the user to operate the Device 13 and communicate with other User's in the system 10. Although the user interface can be designed to have virtually an infinite number of different look and feel implementations, there are certain functions and features that all Devices 13 should have in common. These functions and features are listed below.

**[0193]** The user interface preferably includes one or more of the following status indicators or flags: (i) battery indicator; (ii) connectivity indicator; (iii) clock; (iv) transmitter status; (v) transmission sync status; (vi) Reviewing status; (vii) Priority Messages needing attention; and (viii) missed Messages.

**[0194]** The user interface preferably includes the following functions, flags and components to conduct and manage a single Conversation: (i) name of a Conversation and/or list of Participants; (ii) Conversation status; (iii) Conversation type; (iv) Conversation duration; (v) time behind the Head of the Conversation; (vi) outstanding Messages; (vii) presence/status of Participants; (viii) meta data with navigation; (iix) Conversation attributes or designators; (ix) Conversation set-up including title, scheduling, Participants, Conversation summary; and (v) indicators showing which Participants have contributed Messages and which have Listened to or Reviewed messages.

**[0195]** The user interface also preferably includes, in addition to those listed immediately above, the following functions, flags and components to conduct and manage multiple Conversations: (i) a name/identifier for each Conversation; (ii) live/active or standing/inactive indicator; (iii) Review position, either at Head or Time shifted; (iv) Priority and other attributes; and (v) indicators of what portions of Conversations were missed.

**[0196]** The user interface also preferably includes, a number of navigation features, such as: (i) DVR style fast back/forward per conversation; (ii) instant messaging style personal message navigation; (iii) Conversation Time indicators; (iv) time scale shifting (i.e. zooming backward and forward through a Message or the Messages of a Conversation); (v) changing Priority of Conversations; (vi) hanging up; and (vii) home.

**[0197]** The aforementioned functions and features may be implemented in a variety of ways, for example using a touch-screen graphical user interface 110, or other input devices, such as the data entry buttons, keys, or keyboard 112, a mouse, by voice-activated commands, or a combination thereof. Again the functions and features listed above and how they are implemented is not intended to be exhaustive. The various methods and technologies that could be used is so comprehensive, that it is not practical to list or discuss them all herein.

## J. Conversations

**[0198]** The MCMS application 20 supports a number of different types of Conversations, such as a near real time or "live" calls where the delay from when a Participant speaks and the other Participant(s) hear the first Participant is very small, Conversations where Participants exchange voice Messages back and forth with a longer delay between Messages, "live" conference calls involving multiple Users, standing Conference calls at a regularly scheduled times, or configurable structured call types such as a simultaneous briefing, where Participants each leave a Message briefing beforehand for others to Review to before everyone joins the live conference call. Yet another unique attribute of the MCMS application 20 is the ability for Users to transition between the different types of Conversations. For example, the Participants can seamlessly transition from a voice-messaging mode to a live call and back again. Or the Participants of a live conference call can transition to a voice-messaging mode and send updates or action items to each other after the conference call. While several

examples have been noted, it should be understood that the system 10 is extremely flexible and provides numerous options to transition between different types of calls or Conversations and between multiple conversations. By varying the delay between Messages, the Participants effectively flows between the type of Conversation that best suits their needs. The above examples should therefore not be construed as limiting.

**[0199]** As noted above, Conversations consist of Messages maintained in their original context and sequence. A sent Message either belongs to an existing Conversation or begins a new Conversation. A typical Conversation includes a set of Messages that are organized around a defined subject matter, topic, time, Group or Channel. For example, Conversations may involve a common Group of people such as the members of a club, a company may have a standing Conversation at a fixed time, such as a weekly sales meeting, or friends may have ad-hoc Conversations on various topics, such as making dinner plans.

**[0200]** Each Conversation is defined by a set of attributes, including a name, a list of Participants, begin and end time, and state including at least pending, active, or terminated. Other Conversation states are possible in other embodiments. A User interacts with the MCMS application 20 on their Devices 13. In preferred embodiments, the interface allows a User to organize Conversations by any of the various attributes.

**[0201]** The relationship between a Participant and a Conversation also has attributes. These attributes include, but are not limited to, Priority, Status (states of participation in a Conversation). The values of Participant status include active, participation in more than one Conversation at a time, Reviewing a Conversation in a time-shifted mode, Catching Up To Live, passively participating (i.e., not actively Reviewing, but receiving high Priority Messages), standby, or ignoring a Conversation (i.e., declined to participate or record the Conversation).

**[0202]** From a recipient's perspective, a User may select or define the relative priority of Messages. For example, Messages from a person's boss would typically be given a higher Priority than a social acquaintance. In some embodiments, a recipient has the ability to set their own Priorities. In the implementation of MCMS-C the User selects a subset of their Conversations to be rendered consecutively. The User then sets ordered Priorities for these Conversations. The system uses the priorities set by the user to order the Messages to be rendered. The aforementioned algorithm queues the Messages to be

rendered using User Priorities and information concerning Message data available (beyond MTSD).

**[0203]** In other embodiments, such as tactical communications, the recipient may have no or limited ability to set Priorities. For example, a fire fighter may have no ability to lower the Priority of Messages from the fire chief. A sending-User, however, has the ability to send urgent or highly important Messages. By Tagging a Message as urgent or an emergency, the Message is rendered as soon as possible at the recipient(s), overriding any Priority settings of the recipient. Any conflicts among multiple emergency messages are resolved based on a predetermined priority scheme.

## K. MCMS Operation

**[0204]** Referring to Figure 11A, an organizational diagram 1100 grouping the major functions of the MCMS application 20 is illustrated. The major functions include account management 1102, conversation management 1104, aggregate conversation list management 1106, conversation participation 1108, call controls 1110, and contact management 1112. After registering and logging into the system 10, the User may navigate through the user interface of their Device 13 implementing the various management functions, described in detail below. In some embodiments, this functionality will provide a great deal of flexibility. In other embodiments, such as tactical or communication radios, the implementation of the user interface may be constrained with many of the user functionality and options preconfigured to meet the utility of the device. The discussion here is exemplary and is not intended to be an exhaustive explanation of the MCMS functionality, but rather is intended to provide just an overview of some of the MCMS attributes.

### K.1 Account Management

**[0205]** Under the account management function 1102, a registered User may change certain settings and preferences. A User may change their email address, password, name, phone number, phone password, call-in number, default and/or User defined rendering speeds, Tags, gain or volume levels for Reviewing Messages, Catch Up to Live mode, etc. To make these changes, the User enters the appropriate information through the interface 110 of their Device 13. The MCMS application 20 responds by writing the updated preferences into the MCMS database 22.

### K.2 Conversation Management

**[0206]** As illustrated in Figure 11B, conversation management 1104 is a set of functions that allow a User to view their aggregate Conversation lists, create a new Conversation, update the details of a Conversation, delete a Conversation, and close a Conversation. Each of these functions are described below.

**[0207]** View Conversations 1104a – For each Conversation, the MCMS application 20 may provide the User the one or more of the following attributes: the name of the Conversation, the actual start time, the last activity, Tags, duration, and the list of Participants. For each Participant, the name and/or phone number, status (live, other call, in past, catch-up-mode, offline-reachable, offline-unavailable.)

**[0208]** Create a Conversation 1104b - A User creates a Conversation through the interface 110 by inputting a Conversation name, a list of Contacts, and an optional scheduled start time. If no start time is designated, it is assumed the start time is immediate. In response, the MCMS application 20 creates a new Conversation in the database 22, associating records for each Participant on the Contacts list. The MCMS application 20 also creates in the database 22 Participant records for each User on the Contact list, allowing the caller to receive the presence information of the others on the Contact list. If the Conversation is scheduled, the MCMS application 20 starts the Conversation at the designated time. Otherwise, the Conversation starts right away.

**[0209]** Update Conversation Details 1104c - The User may make changes to a Conversation through the user interface 110. For example, Participants may be added or removed. Any change in status of the participants is updated in the MCMS database 22.

**[0210]** Delete a Conversation 1104d - A User may delete a specific Conversation from their list of Conversations through the interface 110. In response, the MCMS application 20 notes the change in the database 22, and designates the Conversation as deleted.

**[0211]** Close a Conversation 1104e – A User may elect to terminate or close a Conversation. In one embodiment, only the User that creates a Conversation can elect to terminate that Conversation.

### K.3 Aggregate Conversation List Management

**[0212]** As illustrated in Figure 11C, the aggregate conversation list management 1106 is a set of functions that allow a User to engage in multiple Conversations (i.e., the User's aggregate conversation list). The aggregate conversation list management functions

allow a User, through the interface 110 on their device, to participate in one Conversation "live", while participating in other Conversations in a time-shifted mode.

**[0213]** Chose Conversation 1106a – Through the interface 110, a User may select one Conversation among the User's aggregate conversation list as current. The Messages of the current Conversation may be rendered in either the "live" or time-shifted modes. The User may switch the current conversation among the aggregate conversation list from time-to-time.

**[0214]** Switch Conversations Modes1106b – In an optional embodiment, a User may be able to switch from the MCMS, MCMS-C and MCMS-S modes of operation.

### K.4 Conversation Participation

**[0215]** As illustrated in Figure 11D, conversation participation 1108 is a set of functions that allow a User to start a Conversation, receive a notification to join a Conversation, obtain Conversation status information, and hang up on a Conversation.

**[0216]** Start a Conversation 1108a - After a Conversation has been created, either by the User through the interface 110 or the scheduler in the MCMS application, the status of each Participant is checked. If a Participant is offline, then an effort is made to contact that person. If a Participant is online but engaged in another Conversation, then the MCMS application 20 notifies that Participant. The presence status of all online Participants is updated in the database 22.

**[0217]** Receive a Notification 1108b – The system may notify a User that their attention has been requested on a Conversation through a graphic display and/or audible notification via the user interface 110.

**[0218]** Conversation Status 1108c – A user may request the status of a Conversation through the interface 110 of their Device 13. In response, the MCMS application 20 assembles the status information stored in the database 22 and presents the information to the User.

**[0219]** Conversation Pause 1108d– Through user interface 110, a User may hang up or switch away from an active Conversation. In response, the MCMS application 20 updates the User's participant status for active Conversations in the database 22 and directs the Store and Stream module 24 to remove the User from the Conversation.

## K.5 Conversation Controls

**[0220]** As illustrated in Figure 11E, conversation control 1110 is a set of functions that allow a User to control their participation in a Conversation. These functions allow a User to catch-up-to live, skip to the Head, jump to a past location, pause, play faster and play slower when Reviewing the Messages of a Conversation. Each of these functions is triggered by the User through the interface 110 on the Device 13.

**[0221]** Catch-up-to-live 1110a - A User may Catch Up To Live in an ongoing Conversation using the "CTL" function. When this function is activated, the MCMS application 20 checks the last point in the Conversation the User has reviewed and directs the Store and Stream module 24 to render the Messages not previously heard, using a faster than normal rendering option designated by the User, and to seamlessly transition to live mode when it reaches the head.

**[0222]** Jump to Head 1110c – This function enables a User to jump to the Head of a Conversation, skipping over any intervening Messages between the current point of the User in the Conversation and the Head. When implemented, the MCMS application 20 instructs the Store and Stream module to immediately render the Messages at the Head of the Conversation. (If the Head of the Conversation is currently live this is called Jump to Live (JTL).

**[0223]** Jump to past 1110d – This function enables a User to jump back to a previous Message or point in the Conversation, similar to a rewind or replay function. When implemented, the MCMS application 20 instructs the Store and Stream module 24 to render Media starting from the rewind point.

**[0224]** Pause 1110e – This function enables the User to pause the Reviewing of the Messages of a Conversation. In response, the Store and Stream module 24 stops the rendering of Messages until another command is issued.

**[0225]** Play faster 1110f – This function enables a User to render Messages more quickly. In response, the Store and Stream module 24 renders the Messages at a faster rate than normal. The rendering rate may be either specified by the User or the User may select from a number of preset options.

**[0226]** Play slower 1110g - This function enables a User to render Messages more slowly. In response, the Store and Stream module 24 renders the Messages at a slower

rate than normal. The rendering rate may be either specified by the User or the User may select from a number of preset options.

### K.6 Contact Management

**[0227]** As illustrated in Figure 11F, the system 10 provides the User with a host of functions for managing Contact lists and user Groups. These functions include the adding, editing, deleting Contacts and user Groups. Each of these functions are implemented by a User though the interface of their Device 13. Any revisions or deletions in a User's Contact list or Group list is stored in the MCMS database 22.

**[0228]** Add a Contact 1112a – This function enables a User to add a new Contact to their contact list. The new Contact can be either a registered User or an external contact. Typically the name, phone number(s), type of number (cell, office, home, computer, etc.), email address and other personal information are provided for each contact record.

**[0229]** Edit a Contact 1112b – This function enables a User to edit or update an existing contact record.

**[0230]** Delete a Contact 1112c – This function enables a User to remove or delete an existing contact record.

**[0231]** Search for a Contact 1112d – This function enables a User to search for a specific Contact in their contact list. The search may be conducted using a number of criteria, such as name, phone number, most recently called, most frequently called, Group, etc.

**[0232]** Get a Participant list 1112e – This function enables a User to search and retrieve a list of Participants of a Conversation by a number of different search criteria, including for example by name, most recent outgoing calls, most recent incoming calls, most frequently called, etc.

**[0233]** Authorize a caller to review status 1112f – This function enables a first User to authorize other Users to view the first User's status. Non-authorized Users may not view the status of the first User.

**[0234]** Create a Group of Contacts 1112g – This function enables a User to associate a number of Contacts into a Group. When a User defines a Group, the list of Contacts in that Group are stored in the MCMS database 22.

**[0235]** Edit a Contact Group 1112h – This function enables a User to edit a Group, or update the contact information for a member of the Group.

**[0236]** Delete a Contact Group 1112i – This function enables a User to delete a Group.

## L. MCMS Operation

### L.1 MCMS-C

**[0237]** As noted above, the MCMS-C operation is similar to MCMS, with the added feature of enabling Users to manage and participate in multiple conversations consecutively through a hierarchical system of Priorities and the time-shifting of Messages, which are automatically managed by the system. Implementing the MCMS-C functionality includes three basic processes. As illustrated in Figure 12A, the first process involves defining a set of Conversations for consecutive rendering. Once the list is defined, a hierarchical set of Priorities and other factors are applied to the indexed media payloads associated with the set of Conversations. The indexed media payloads are then sequenced into a sequencing order. By rendering the media in the sequence order, the Messages of the set of Conversations are consecutively rendered.

**[0238]** Referring to Figure 12A, a flow chart illustrating the steps for defining the list of Conversations to render consecutively is shown. In the initial step 1202, the User's aggregate list of Conversations is defined. Either the User or pre-configuration data (step 1204) is next used to select the Conversations among the aggregate list for consecutive rendering (step 1206). With a tactical communication system for example, typically pre-configuration data is used to impose the Conversations to be consecutively rendered. With non-tactical applications, the User is typically given a higher degree of flexibility to select the Conversations for consecutive rendering.

**[0239]** Referring to Figure 12B, a flow chart illustrating the steps for defining hierarchical set of priorities for rendering the Messages of consecutive Conversations is illustrated. In the initial step (1208), a set of priority rules are defined and applied to the list of Conversations to be rendered consecutively (1206). In various embodiments, the set of priority rules may range from a rigid, hierarchical communication protocol to a highly flexible communication protocol. For example in a tactical communication system where a rigid hierarchy is often desirable, the set of priority rules will preferably impose a specific order in which concurrent Messages are rendered. For example with a first-responder tactical system, Messages from a fire chief may be given the highest

priority. The next level of priority may be given to fire fighters inside a burning building. At the next level, priority may be given to fire fighters outside the building, etc. By defining a rigid priority, the current Messages of those supervising the efforts to fight the fire, or those in harms way, are rendered ahead of those performing a less critical role. With non-tactical communications, a User may be given a great deal of flexibility to define their own priority scheme to meet personal needs. A sales executive may for example define a priority scheme listing consecutive Conversations with clients from the most important to the least important. Or a User may prioritize consecutive Messages among family and friends. Regardless of the scheme used, a priority list for consecutive Conversations is defined (step 1210) in this process.

**[0240]** Referring to Figure 12C, a flow diagram illustrating the construction of a queue of Messages received from the various consecutive Conversations is illustrated. In the initial step, the availability of non-rendered indexed media payloads of Messages (i.e., media streams) is continually detected (step 1212) for each of the Conversations to be consecutively rendered. The priority hierarchy is applied to the available indexed media payload streams (step 1214). Based at least partially on the priority hierarchy, and possible other parameters as mentioned below, the available indexed media payloads are arranged into a sequence order (step 1216). The indexed media payloads are then continuously rendered in the sequence order (step 1218). By continuously repeating the above-described process, the Messages of multiple Conversations are consecutively rendered.

**[0241]** In one embodiment, the sequencing order is based either partly or fully on the priority hierarchy. In alternative embodiments, other parameters in addition to the hierarchy and availability may be considered as well. For example, the sequence order may be defined using one or more parameters such as the switching costs associated with interrupting a currently rendered stream of indexed media payloads with the indexed media payloads of a higher priority Conversation, the quality of the available streams of indexed media payloads, the relative time the indexed media payloads were received, a shuffling order, or from input of a controller of the system.

**[0242]** Typically when conflicts between the Messages of different Conversations occur, the indexed media payloads first in sequence order are rendered while the rendering of other available indexed media payloads are paused or delayed. When there

no conflicts, indexed media payloads are immediately rendered as they become available.

**[0243]** In yet another embodiment, the Messages of the consecutively rendered Conversations may be optionally reviewed in a time-shifted mode. If the User of the first communication device generated any Media associated with the consecutively rendered Conversations, that media is indexed and stored in the PIMB 30 of the Device as well as the PIMB(s) 85 of the Servers 16 on the network. Thus when a Conversation is reviewed in the time shifted mode, the User has the option of reviewing just the incoming Messages associated with the Conversation, or both the incoming Messages as well as the Media created by the first User associated with the Conversation in time index order.

### L.2 MCMS-S Operation

**[0244]** In the MCMS-S or simultaneous mode, a User of a Client 12 enabled Device 13 may define a set of Conversations for simultaneous rendering. Once the set of Conversations is defined, the indexed media payload streams associated with the set of Conversations are simultaneously rendered on the Device 13, regardless if they overlap or not. In alternative embodiments, the User may optionally render the received indexed media payloads from the set of media streams separately. The indexed media payloads of the simultaneous Conversations may also be optionally rendered in near real-time or in the time-shifted mode.

### L.3 MCMS, MCMS-C and MCMS-S Examples

**[0245]** In Figures 13A through 13D, a series of diagrams illustrating the attributes of a Conversation and the operation of MCMS, MCMS-C and MSMS-S is illustrated.

**[0246]** In Figure 13A, a time diagram illustrating the sequence of rendering the indexed media payloads of the Messages of a Conversation labeled "A" between a User "X" and two other users, designated "Y" and "Z". In this example, Media is generated by user Y during the time intervals designated by t1, t5, 56, t7 and t9. Media is generated by user Z during the time intervals designated t3, t6 and t9 through t10.

**[0247]** The rendering sequence at the Device 13 of User X is illustrated at the bottom of the figure. During intervals t1, t5, and t7, Media derived only from Y is rendered. During intervals t3 and t10, only Media derived from Z is rendered. In intervals t6 and t9, Media from both Y and Z is rendered. In intervals t2, t4 and t8, nothing is being

rendered because neither users Y or Z are generating Media during these periods. It should be noted that intervals t1 through t10 are not intended to mean fixed time intervals, but rather, just periods of time when Media is being generated.

**[0248]** The diagram of Figure 13A is useful in illustrating the attributes of Conversations. When one user is generating Media (either Y or Z), that Media is received at the Device 13 of X and is available for rendering. When both users X and Y are generating Media, both Media streams are received at Device 13 of X and is available for Mixing. When neither user X or Y is generating Media, no Media is received at Device 13 of X. As noted above, User X has the option of Reviewing the Media generated during Conversation A in either the near real-time or time-shifted modes. Also the User X has the option of Reviewing the Media in the Mixed format as illustrated, or to separately Reviewing the Media from Y and Z in the time-shifted mode.

**[0249]** Figure 13B illustrates the operation of MCMS. In this example, the User is Participating in three Conversations, designated A, B and C. For Conversations A, B, and C, the User either generates or receives Messages designated (A1, A2, A3, and A4), (B1, B2 and B3) and (C1and C2) respectively. The timing and duration of each Message is indicated by the starting point along the time-line. It is useful to note that in this example, all of the Messages overlap in time to some degree, except Message B2.

**[0250]** With the MCMS application, the User selects one Conversation as current. For the selected Conversation, the User may Review incoming Messages and generate Messages which are transmitted to the other Participants of the Conversation. In this example, the User selects in order Conversation B, C and A as current respectively. The Message sequence is therefore initially B1, B2, and B3, followed by C1 and C2, and then finally A1 through A4. Again, while a particular Conversation is selected as current, the User may transition between the near real-time and the time-shifted modes and back again. The final rendering as illustrated in the diagram is not intended to correspond in timing of the received Messages as illustrated in the top portion of the diagram. Rather the lower portion of the diagram is intended to show only the rendering order of the Messages, based on Conversation order selected by the User.

**[0251]** The example of Figure 13B is thus useful in illustrating the attributes of the MCMS application. A User selects one Conversation as current. The other

Conversations are paused. The User may also transition the current Conversation among all the Conversations at any time.

**[0252]** Referring to Figure 13C, a diagram illustrating the operation of MCMS-C is illustrated. In this example, the User is Participating in two consecutive Conversations, A and B. With Conversation A, three Messages are received A1, A2, and A3. With Conversation B, three Messages are received B1, B2 and B3. It is useful to note that with this example, Message B1 conflicts with Messages A1. Also Conversation A has a higher priority that Conversation B.

**[0253]** During the consecutive rendering of the two Conversations, the higher priority Messages A1 and A2 are first rendered in near real-time, as represented by the vertical dashed lines in the figure. Since there is a relatively large time gap between Messages A2 and A3, this space is filled by time shifting and rendering Messages B1 and B2. When A3 arrives, it is rendered in near real-time, while Message B3 is rendered only after the higher priority Message A3 is rendered. By time-shifting rendering of the lower priority Messages between the higher Priority Messages, consecutive multiple conversations can be managed. It should be noted that in this simple example, priority is the only parameter used to determine the consecutive order for rendering. As noted above, a number of other parameters may be used as well.

**[0254]** Referring to Figure 13D, a diagram illustrating MCMS-S is illustrated. In this example, a User is engaged in three simultaneous Conversations, A, B and C. The Messages A1, A2 and A3, B1, B2 and B3, and C1 and C2 are received for each Conversation A, B and C are shown in the diagram respectively. With the MCMS-S, the incoming Messages are rendered at a recipients Device 13 as they are received. The rendering sequence of the Messages of the three Conversations A, B and C, as illustrated in the lower portion of the diagram, are therefore the same as when the Messages were received. In this manner, multiple Conversations may be simultaneously rendered.

**[0255]** In the aforementioned examples, several variations of the MCMS application were described, including MSMS-C and MCMS-S. Regardless of the specific type of MCMS application used, they all share several common characteristics. In each case, the Conversations are defined by a threaded sequence or organization of Messages. The Messages are segmented from a stream of Media, with each Message given a sequence

identifier and indexed by the time the Media was created. Depending on the variation of the MCMS application, Messages can be rendered in accordance with one or more rendering options. The rendering options include, in one form or another, the filtering, grouping overlapping and/or serialization of Messages, using anywhere from zero to a plurality of different attributes. In this manner, multiple Conversations, each including a string of Messages, can be conducted on a single Client 12 enabled Device 13. Lastly, each of the variations of MCMS may handle the receipt of interrupt Messages in the same way. When an interrupt Message is received, it typically takes precedent over and is rendered before other Messages belonging to other Conversations.

## M. Client and Server Hardware

**[0256]** Referring to Figure 14A, a block diagram 140 illustrating the hardware of a Device 13 used for storing and executing the Client application 12 is shown. The hardware includes a CPU 142, main memory 144 and mass storage 146. As is well known in the art, the Client application 12 is loaded and stored in main memory144 and mass storage 146 and executed by the CPU 142.

**[0257]** Referring to Figure 14B, a block diagram 150 illustrating the hardware of a Server 16 used for storing and executing the server application 78 is shown. The hardware includes a CPU 152, main memory 154, mass storage 156, and the archive 89. As is well known in the art, the server application 78 is loaded and stored in main memory 154 and mass storage 156 and executed by the CPU 152. As noted above, indexed media payloads of one or more Users are stored in the archive 89.

**[0258]** Although many of the components and processes are described above in the singular for convenience, it will be appreciated by one of skill in the art that multiple components and repeated processes can also be used to practice the techniques of the system and method described herein. Further, while the invention has been particularly shown and described with reference to specific embodiments thereof, it will be understood by those skilled in the art that changes in the form and details of the disclosed embodiments may be made without departing from the spirit or scope of the invention. For example, embodiments of the invention may be employed with a variety of components and should not be restricted to the ones mentioned above. It is therefore intended that the invention be interpreted to include all variations and equivalents that fall within the true spirit and scope of the invention.

## CLAIMS

What is claimed is:

1. A communication method for communication on a first communication device, comprising:

receiving at the first communication device media of a conversation with one or more participants over a network;

storing on the first communication device the received media as the media is received over the network;

rendering on the first communication device the received media of the conversation in either a near real-time mode or a time-shifted mode; and

transitioning the rendering of the media of the conversation on the first communication device from the time-shifted mode to the near real-time mode by:

retrieving the stored media;

rendering the retrieved media from storage at an increased rendering rate;

ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the conversation as it is being received over the network; and

rendering the media of the conversation in the near real-time mode as it is being received over the network after the catch-up point.

2. The method of claim 1, wherein retrieving the stored media further comprises retrieving the stored media from a selected previous point in time in the conversation.

3. The method of claim 2, wherein rendering the retrieved media from storage further comprises rendering the stored media from the selected previous point in time in the conversation.

4. The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media at a rendering rate faster than the media was originally encoded.

5. The method of claim 4, wherein the rendering rate is variable.

6. The method of claim 5, wherein the variable rendering rate is selected at the first communication device.

7. The method of claim 1, wherein rendering the media of the conversation in the near real-time mode as it is being received over the network further comprises progressively rendering the received media as the media arrives over the network.

8. The method of claim 1, wherein rendering the received media of the conversation in the time-shifted mode further comprises:

retrieving the retrieved media from storage after a delay; and

rendering the media retrieved from storage after the delay.

9. The method of claim 1, wherein the rendering on the first communication device may be selectively transitioned between the near real-time mode and the time-shifted mode.

10. The method of claim 1, further comprising providing a plurality of rendering options for rendering the media, the rendering options including one or more of the following: pause, replay, play faster, play slower, jump backward, jump forward, catch up to the most recently received media, or jump to the most recently received media.

11. The method of claim 1, further comprising:

creating media on the first communication device, the created media associated with the conversation;

storing the created media; and

transmitting the created media to the one or more participants of the conversation over the network.

12. The method of claim 11, further comprising segmenting the created and the received media into messages.

13. The method of claim 12, further comprising associating the messages with the conversation.

14. The method of claim 1, further comprising mixing the media from the one or more participants of the conversation when rendering in the near real-time mode.

15. The method of claim 1, further comprising mixing the media from the one or more participants of the conversation when rendering in the time-delayed mode.

16. The method of claim 1, further comprising separating the media from the one or more participants of the conversation when rendering in the time-delayed mode so that the media from each participant of the conversation is independently rendered.

17. The method of claim 1, wherein the media comprises one or more of the following: voice, video, text, sensor data, position or GPS information, or a combination thereof.

18. The method of claim 1, wherein the first communication device comprises one of the following: a programmable land-line phone, a programmable wireless or cellular phone, a programmable computer or a programmable radio, a satellite phone or radio, or a programmable tactical radio or phone.

19. The method of claim 1, wherein the network comprises a packet-based network, a circuit-based network, a cellular network, a wireless network, a radio based network, or a combination thereof.

20. The method of claim 1, wherein the media comprises one or more indexed media payloads each having a time stamp and sequence number.

21. The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media

by performing one or more of the following: skipping silence in the media, skipping hesitation particles in the media, speeding up the audio content in the media, or pitch shifting.

22. The method of claim 8, wherein the delay has a duration, the duration being determined by a user of the first communication device.

## ABSTRACT OF THE DISCLOSURE

A telecommunication and multimedia management apparatus and method that supports voice and other media communications and that enables users to: (i) participate in multiple conversation modes, including live phone calls, conference calls, instant voice messaging or tactical communications; (ii) review the messages of conversations in either a live mode or a time-shifted mode and to seamlessly transition back and forth between the two modes; (iii) participate in multiple conversations either concurrently or simultaneously; (iv) archive the messages of conversations for later review or processing; and (v) persistently store media either created or received on the communication devices of users. The latter feature enables users to generate or review media when either disconnected from the network or network conditions are poor and to optimize the delivery of media over the network based on network conditions and the intention of the users participating in conversations.

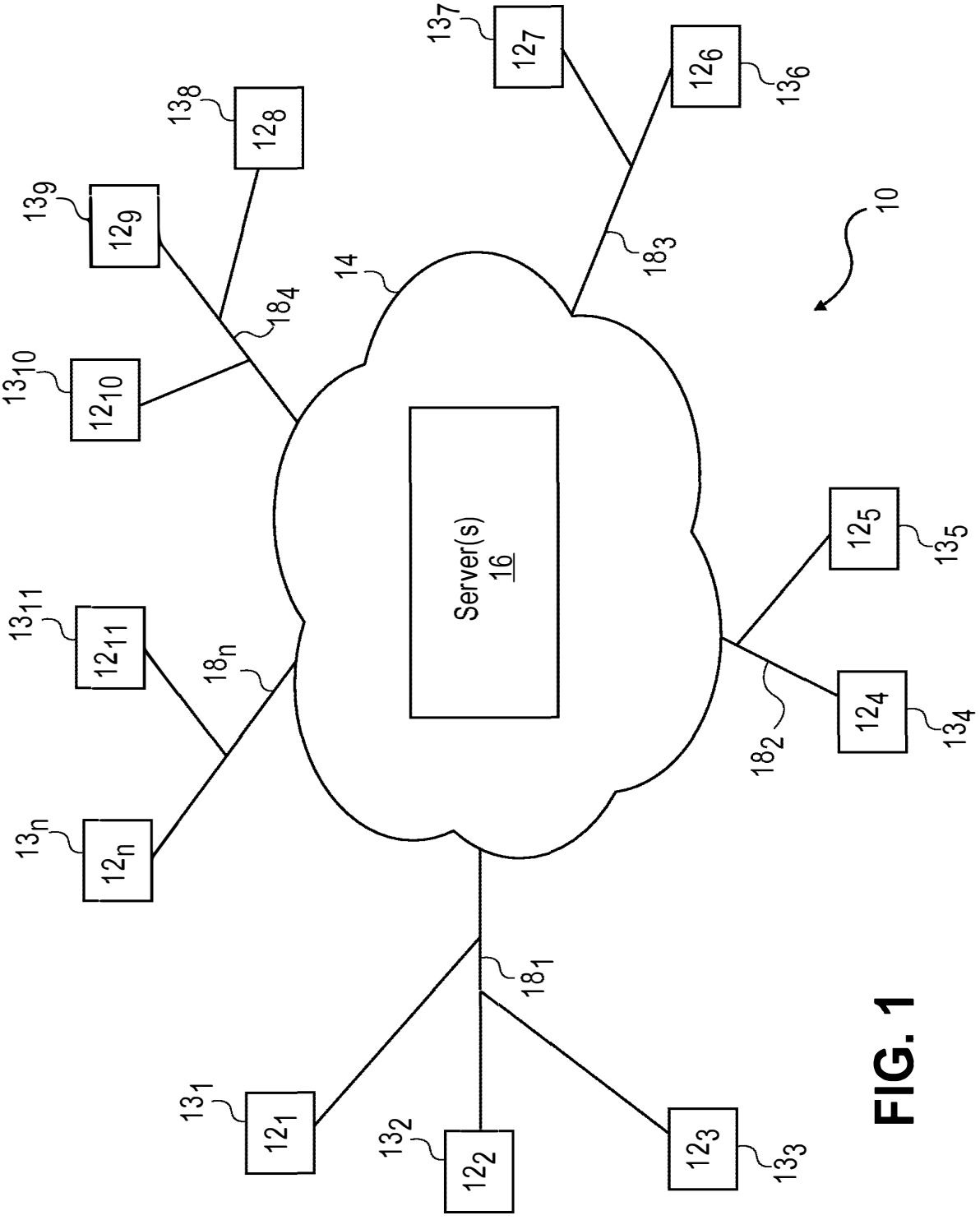

Server(s)
16
14
10
$12_1$
$12_2$
$12_3$
$12_4$
$12_5$
$12_6$
$12_7$
$12_8$
$12_9$
$12_{10}$
$12_{11}$
$12_n$
$13_1$
$13_2$
$13_3$
$13_4$
$13_5$
$13_6$
$13_7$
$13_8$
$13_9$
$13_{10}$
$13_{11}$
$13_n$
$18_1$
$18_2$
$18_3$
$18_4$
$18_n$

FIG. 1

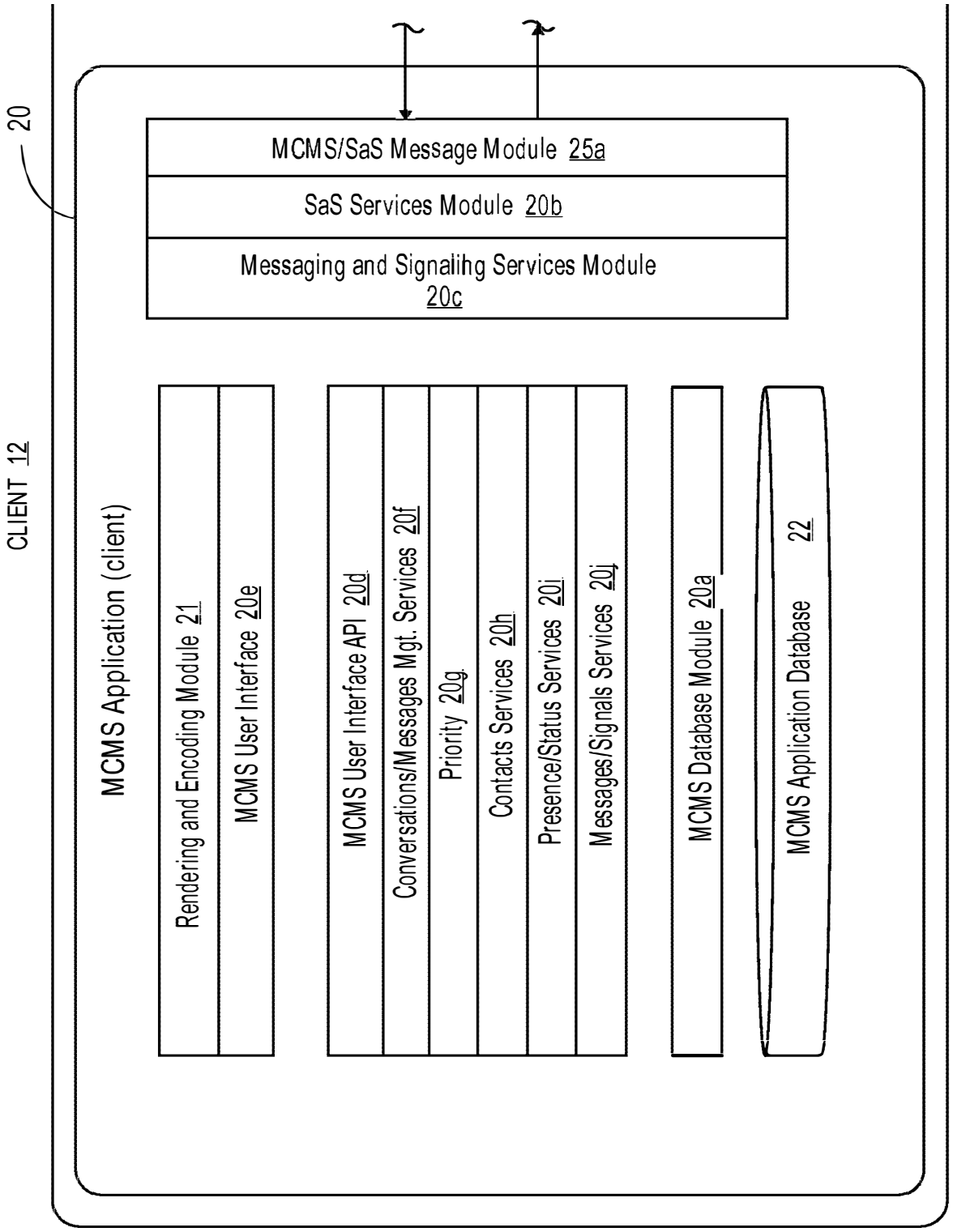



FIG. 2A

FIG. 2A | FIG. 2B

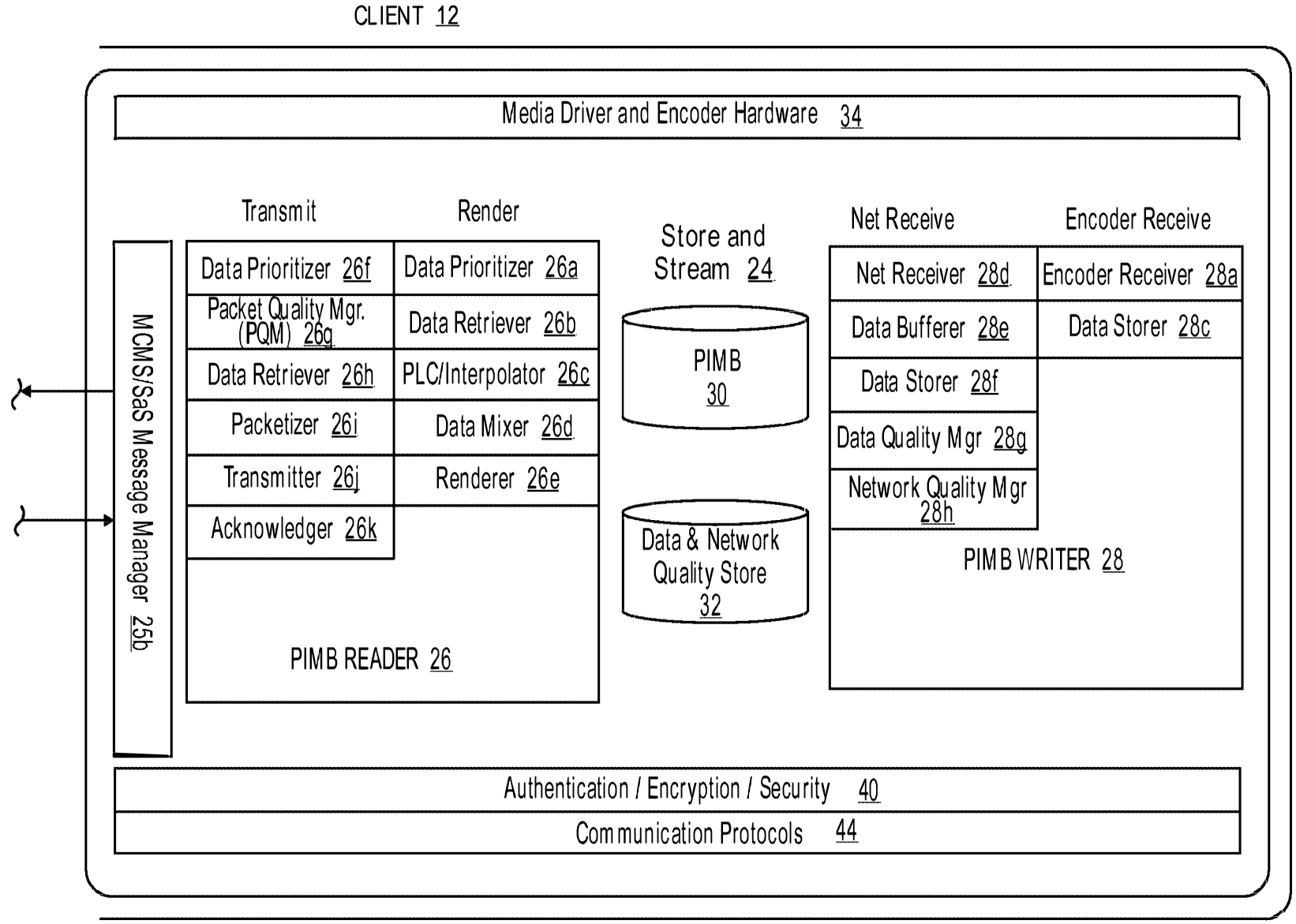



FIG. 2B

| FIG. 2A | FIG. 2B |
|---|---|

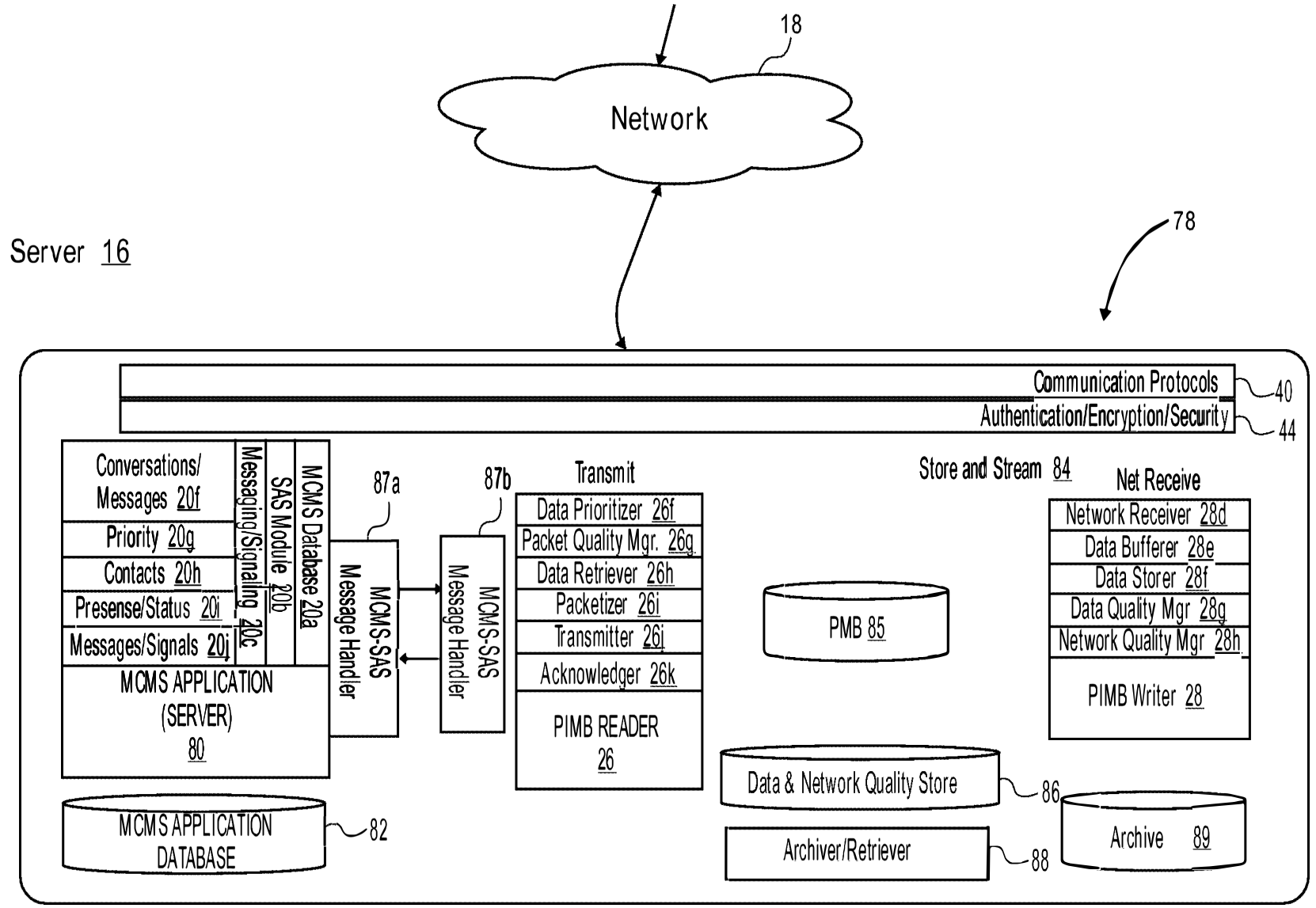

18
Network
Server 16
78
Communication Protocols
40
Authentication/Encryption/Security
44
Conversations/ Messages 20f
Priority 20g
Contacts 20h
Presense/Status 20i
Messages/Signals 20j
Messaging/Signaling 40c
SAS Module 20b
MCMS Database 20a
MCMS APPLICATION (SERVER) 80
87a
MCMS-SAS Message Handler
87b
MCMS-SAS Message Handler
Transmit
Data Prioritizer 26f
Packet Quality Mgr. 26g
Data Retriever 26h
Packetizer 26i
Transmitter 26j
Acknowledger 26k
PIMB READER 26
Store and Stream 84
PMB 85
Net Receive
Network Receiver 28d
Data Bufferer 28e
Data Storer 28f
Data Quality Mgr 28g
Network Quality Mgr 28h
PIMB Writer 28
MCMS APPLICATION DATABASE
82
Data & Network Quality Store
86
Archiver/Retriever
88
Archive 89

FIG. 3

083

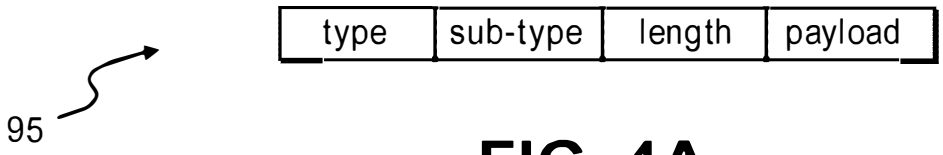



FIG. 4A

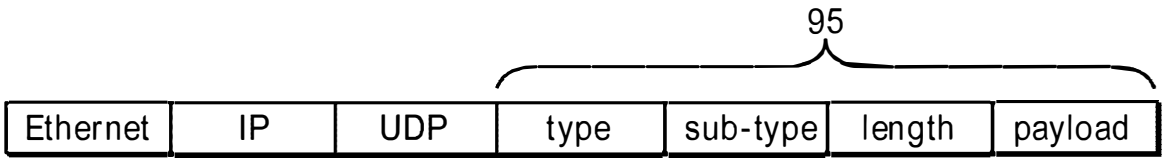



FIG. 4B

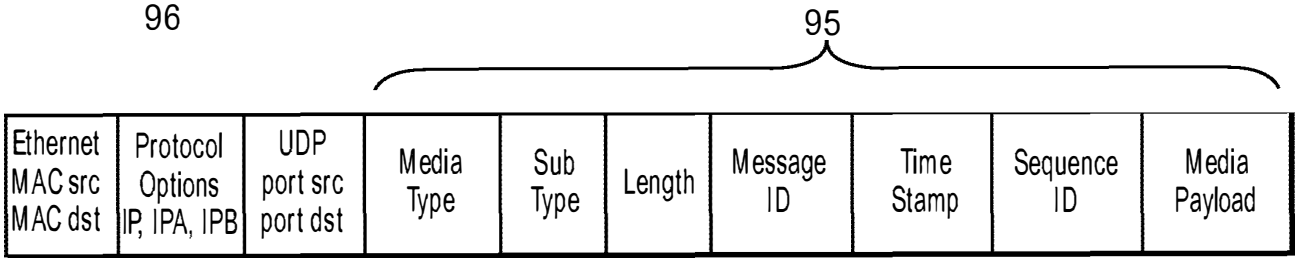



FIG. 4C

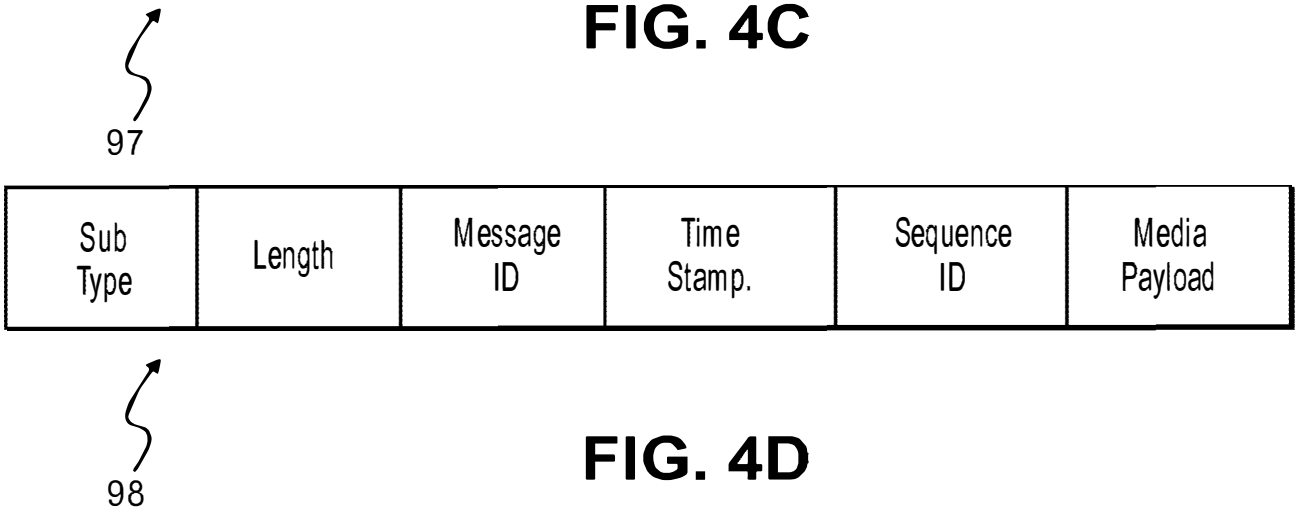



FIG. 4D

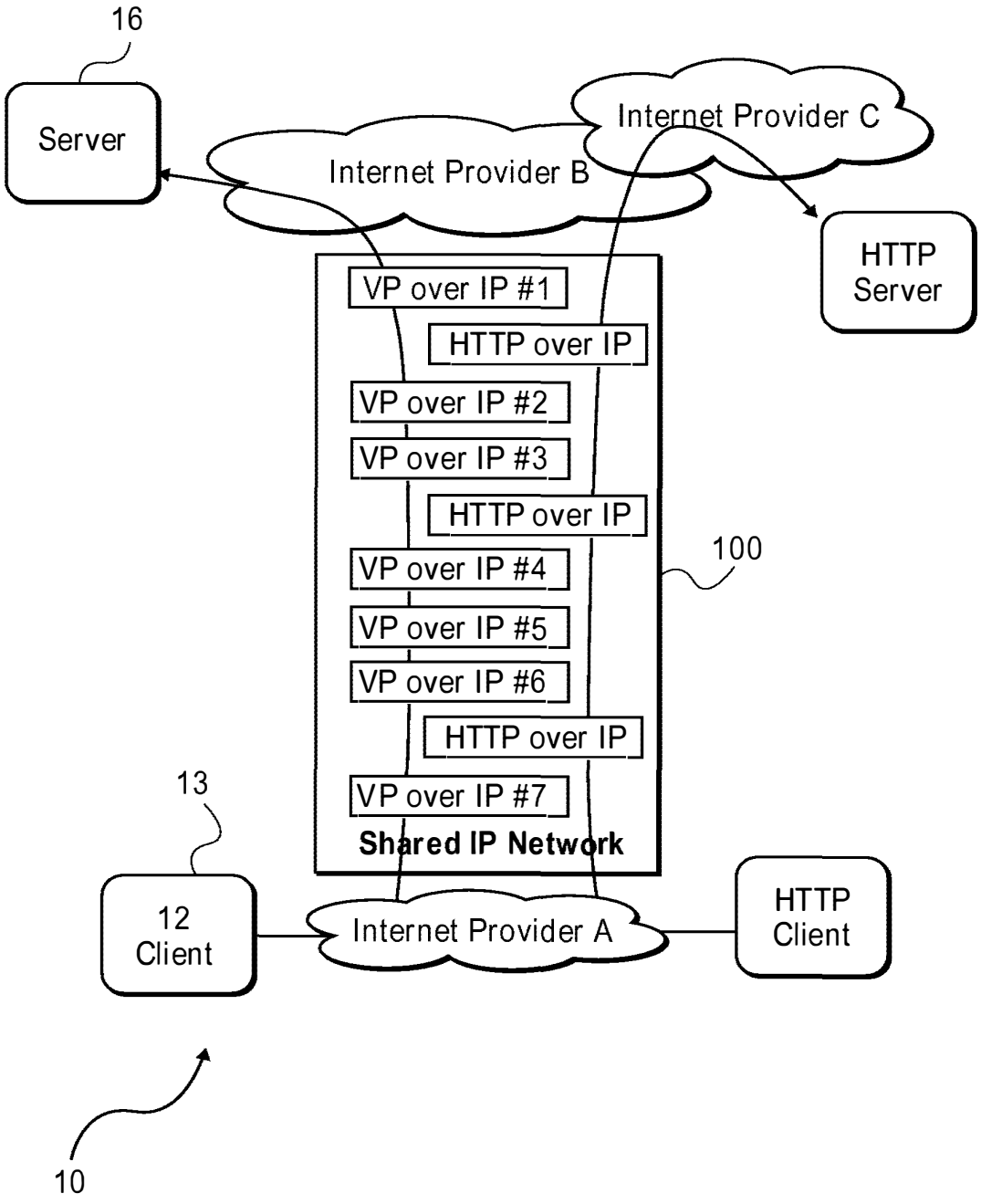

16
Server
Internet Provider C
Internet Provider B
HTTP
Server
VP over IP #1
HTTP over IP
VP over IP #2
VP over IP #3
HTTP over IP
100
VP over IP #4
VP over IP #5
VP over IP #6
HTTP over IP
VP over IP #7
Shared IP Network
13
12
Client
Internet Provider A
HTTP
Client
10

FIG. 5

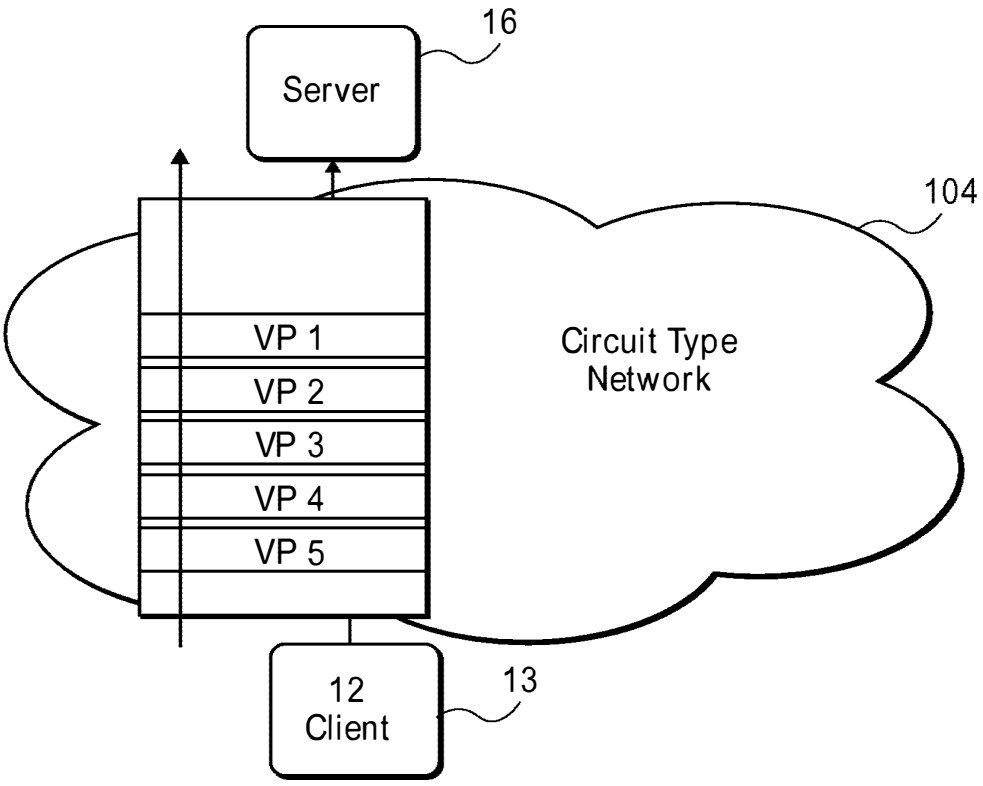

16
Server
104
VP 1
VP 2
VP 3
VP 4
VP 5
Circuit Type
Network
12
Client
13

FIG. 6

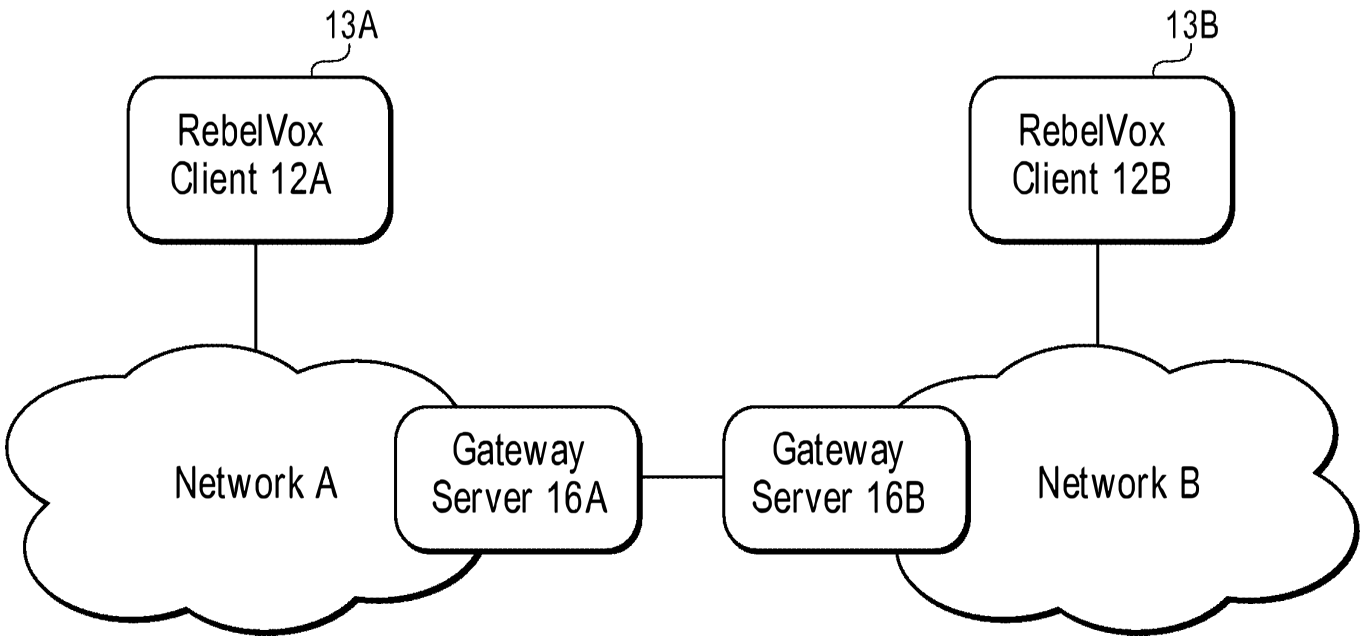

13A
13B
RebelVox
Client 12A
RebelVox
Client 12B
Network A
Gateway
Server 16A
Gateway
Server 16B
Network B

FIG. 7

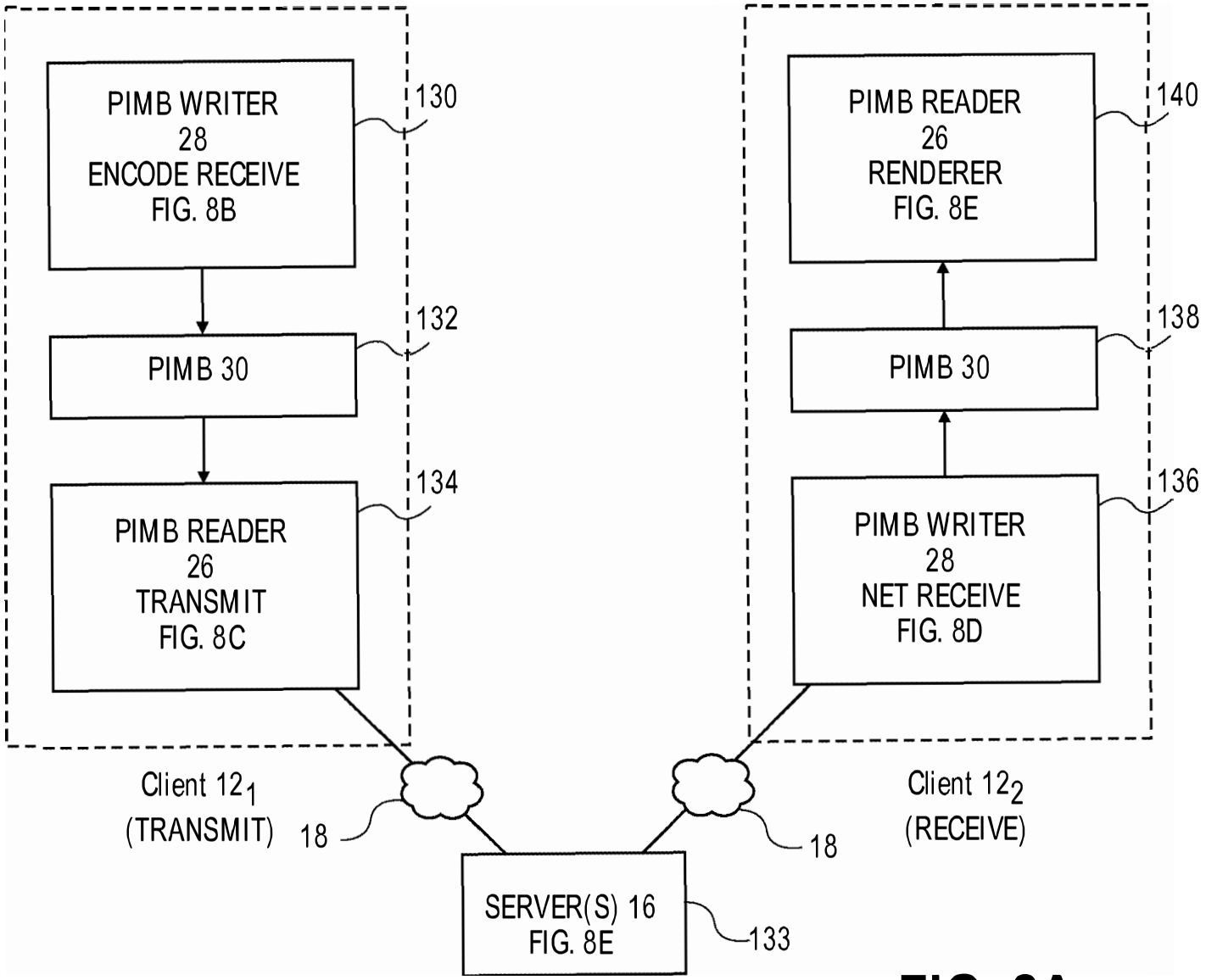

PIMB WRITER
28
ENCODE RECEIVE
FIG. 8B
130
PIMB 30
132
PIMB READER
26
TRANSMIT
FIG. 8C
134
Client 12$_1$
(TRANSMIT)
18
PIMB READER
26
RENDERER
FIG. 8E
140
PIMB 30
138
PIMB WRITER
28
NET RECEIVE
FIG. 8D
136
Client 12$_2$
(RECEIVE)
18
SERVER(S) 16
FIG. 8E
133

FIG. 8A

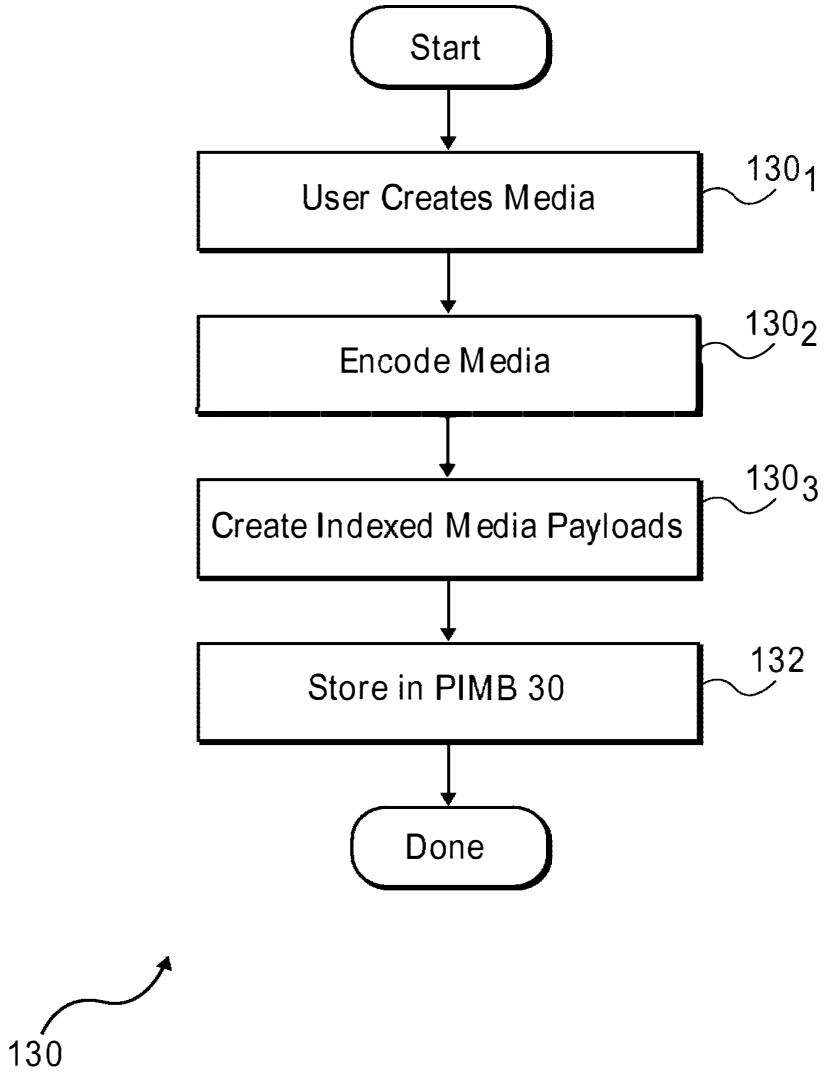

Start
User Creates Media
$130_1$
Encode Media
$130_2$
Create Indexed Media Payloads
$130_3$
Store in PIMB 30
132
Done
130

FIG. 8B

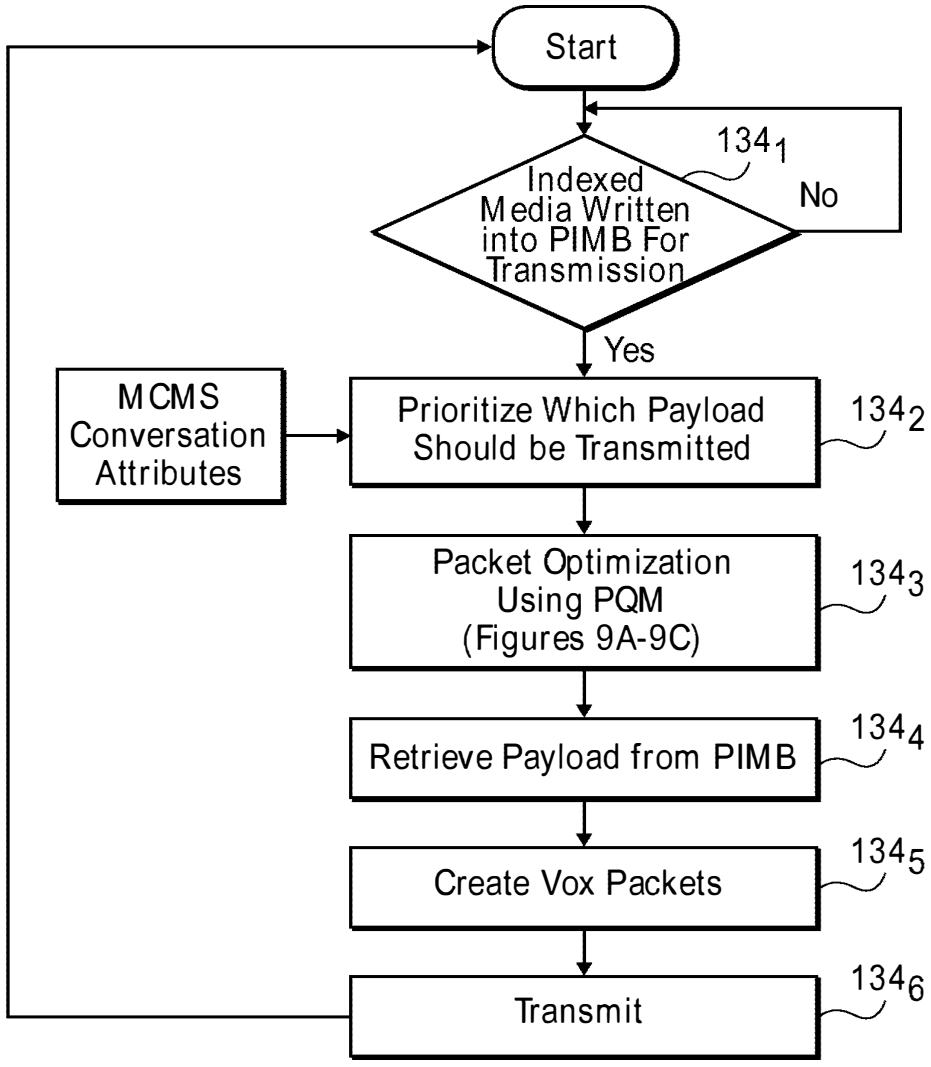

Start
Indexed Media Written into PIMB For Transmission
1341
No
Yes
MCMS Conversation Attributes
Prioritize Which Payload Should be Transmitted
1342
Packet Optimization Using PQM (Figures 9A-9C)
1343
Retrieve Payload from PIMB
1344
Create Vox Packets
1345
Transmit
1346

FIG. 8C

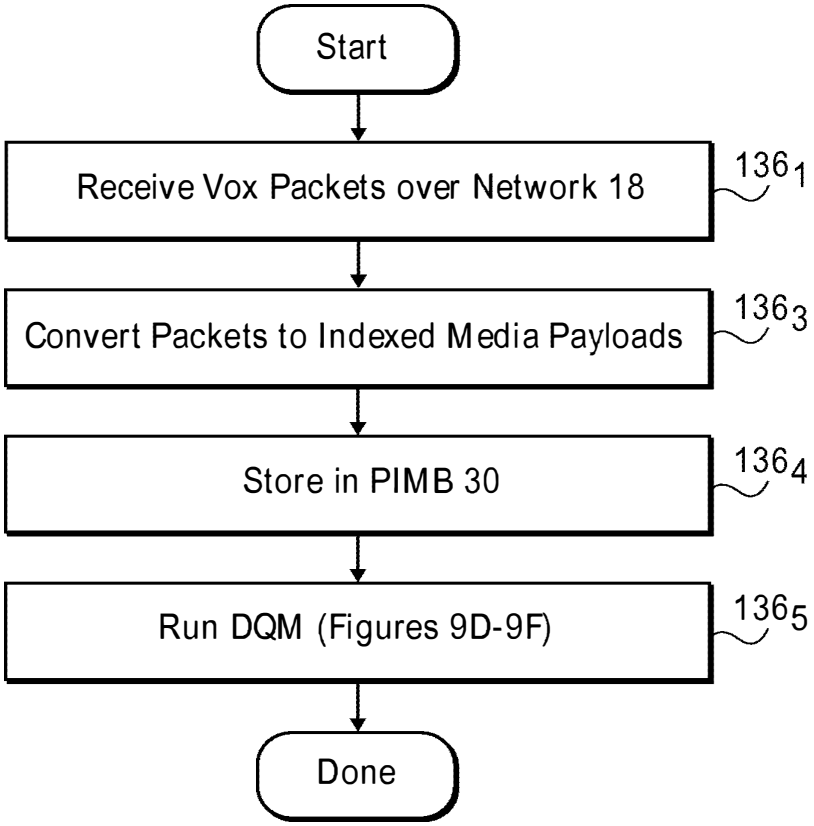

Start
Receive Vox Packets over Network 18
136 1
Convert Packets to Indexed Media Payloads
136 3
Store in PIMB 30
136 4
Run DQM (Figures 9D-9F)
136 5
Done

FIG. 8D

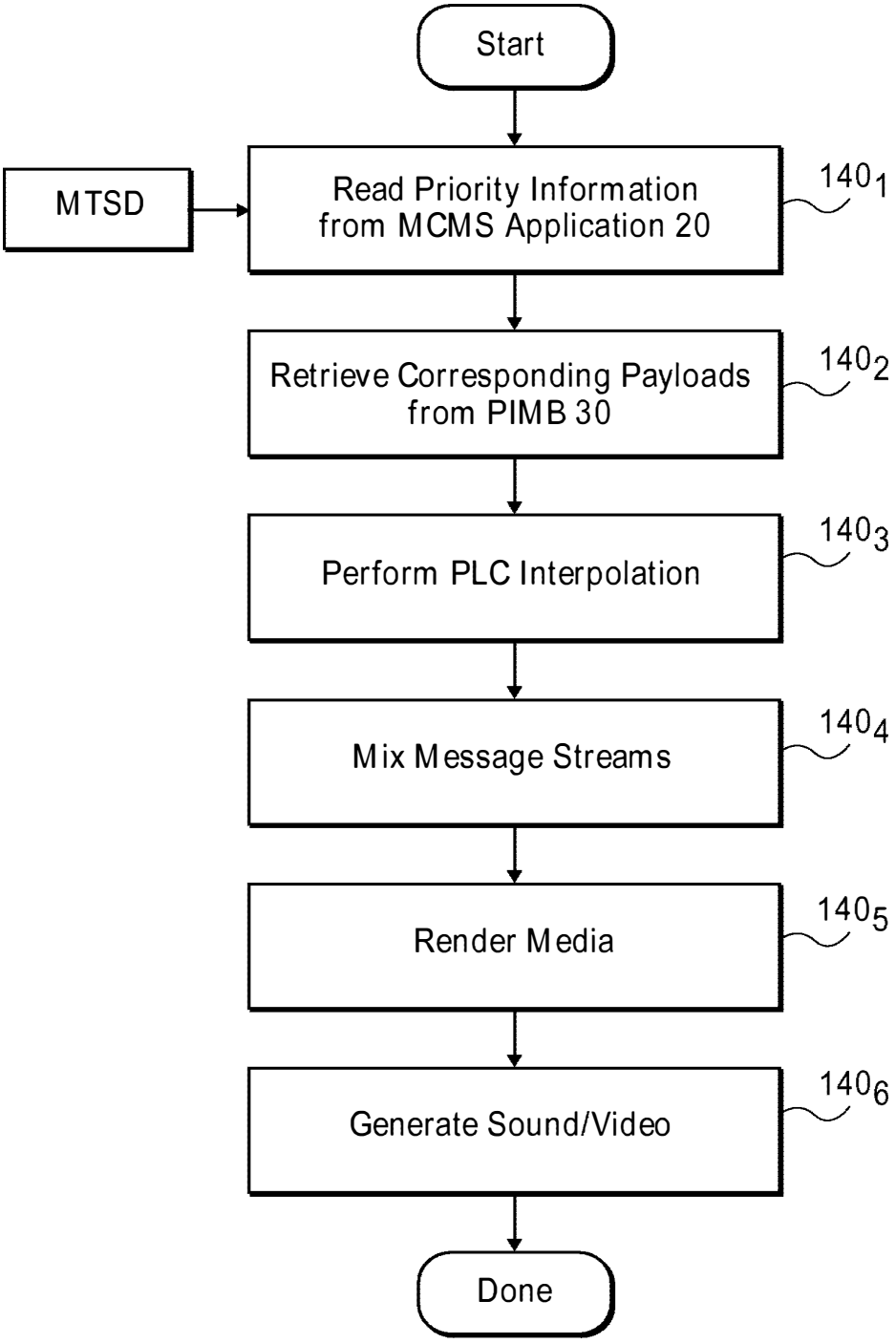

Start
MTSD
Read Priority Information from MCMS Application 20
$140_1$
Retrieve Corresponding Payloads from PIMB 30
$140_2$
Perform PLC Interpolation
$140_3$
Mix Message Streams
$140_4$
Render Media
$140_5$
Generate Sound/Video
$140_6$
Done

FIG. 8E

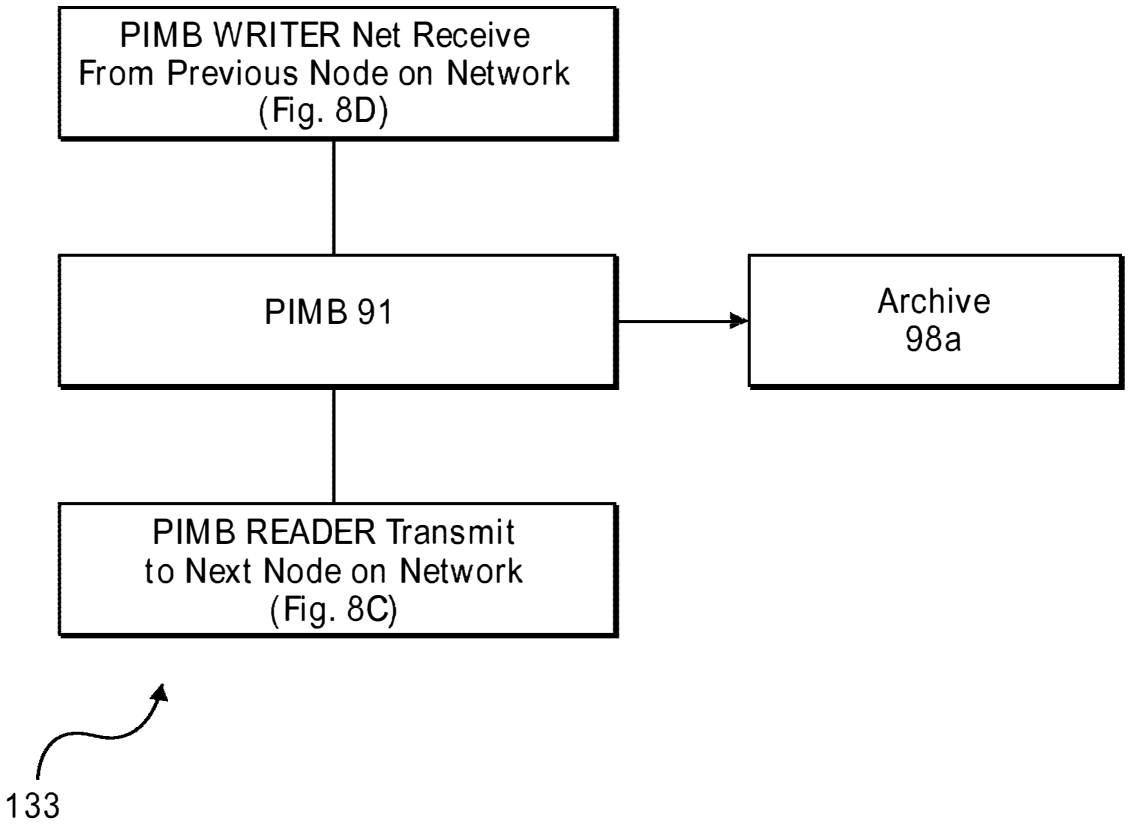

PIMB WRITER Net Receive
From Previous Node on Network
(Fig. 8D)
PIMB 91
Archive
98a
PIMB READER Transmit
to Next Node on Network
(Fig. 8C)
133

FIG. 8F

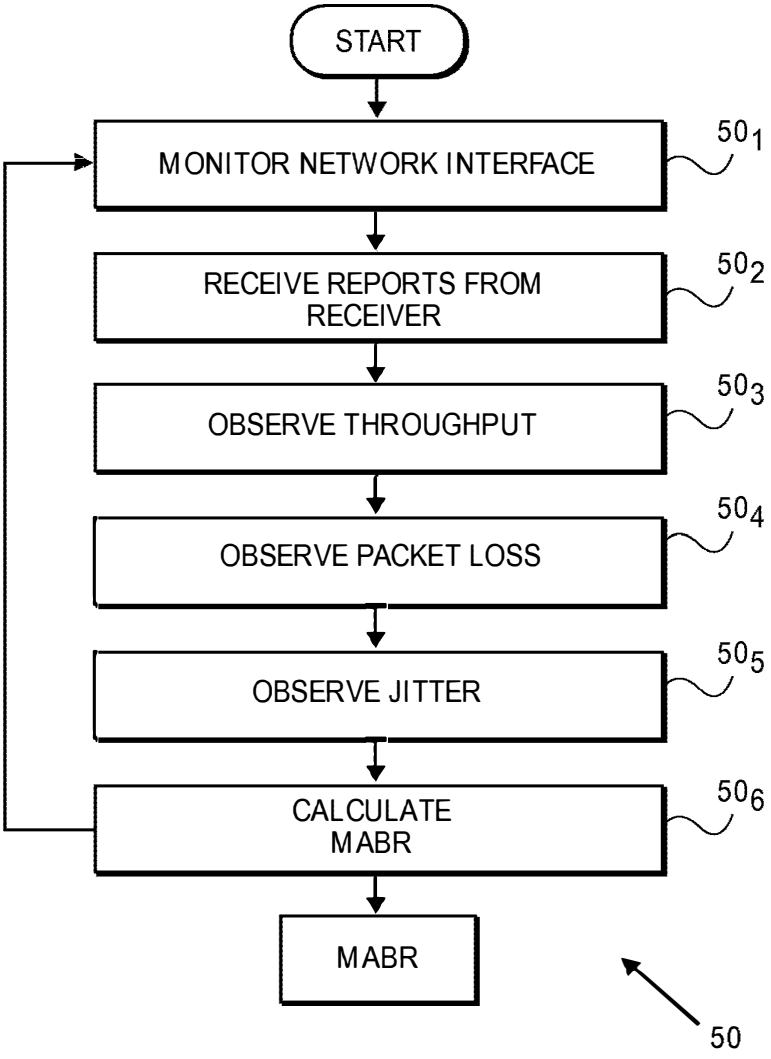

START
MONITOR NETWORK INTERFACE
501
RECEIVE REPORTS FROM RECEIVER
502
OBSERVE THROUGHPUT
503
OBSERVE PACKET LOSS
504
OBSERVE JITTER
505
CALCULATE MABR
506
MABR
50

FIG. 9A

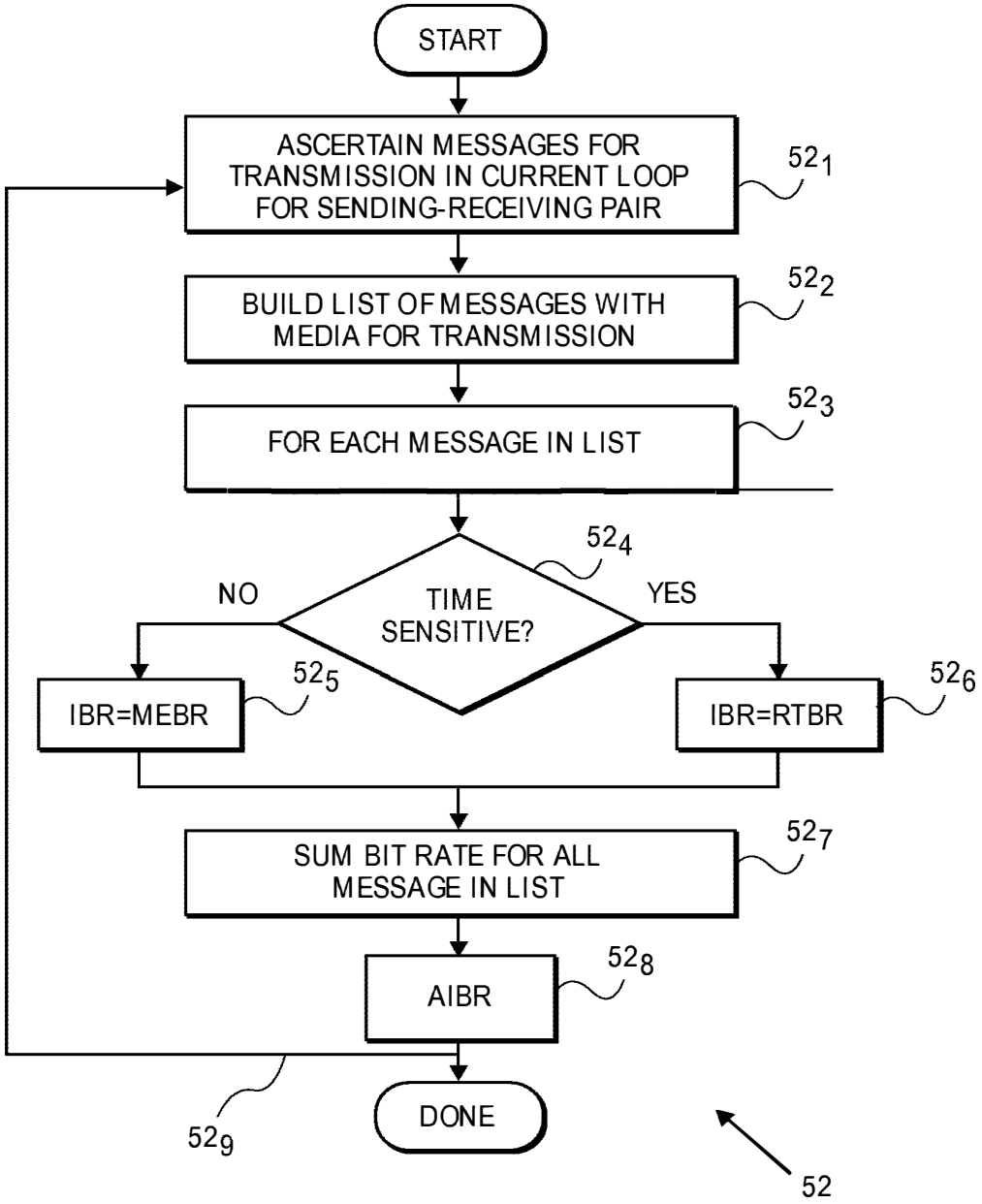

START
ASCERTAIN MESSAGES FOR TRANSMISSION IN CURRENT LOOP FOR SENDING-RECEIVING PAIR
521
BUILD LIST OF MESSAGES WITH MEDIA FOR TRANSMISSION
522
FOR EACH MESSAGE IN LIST
523
524
TIME SENSITIVE?
NO
YES
IBR=MEBR
525
IBR=RTBR
526
SUM BIT RATE FOR ALL MESSAGE IN LIST
527
AIBR
528
DONE
529
52

FIG. 9B

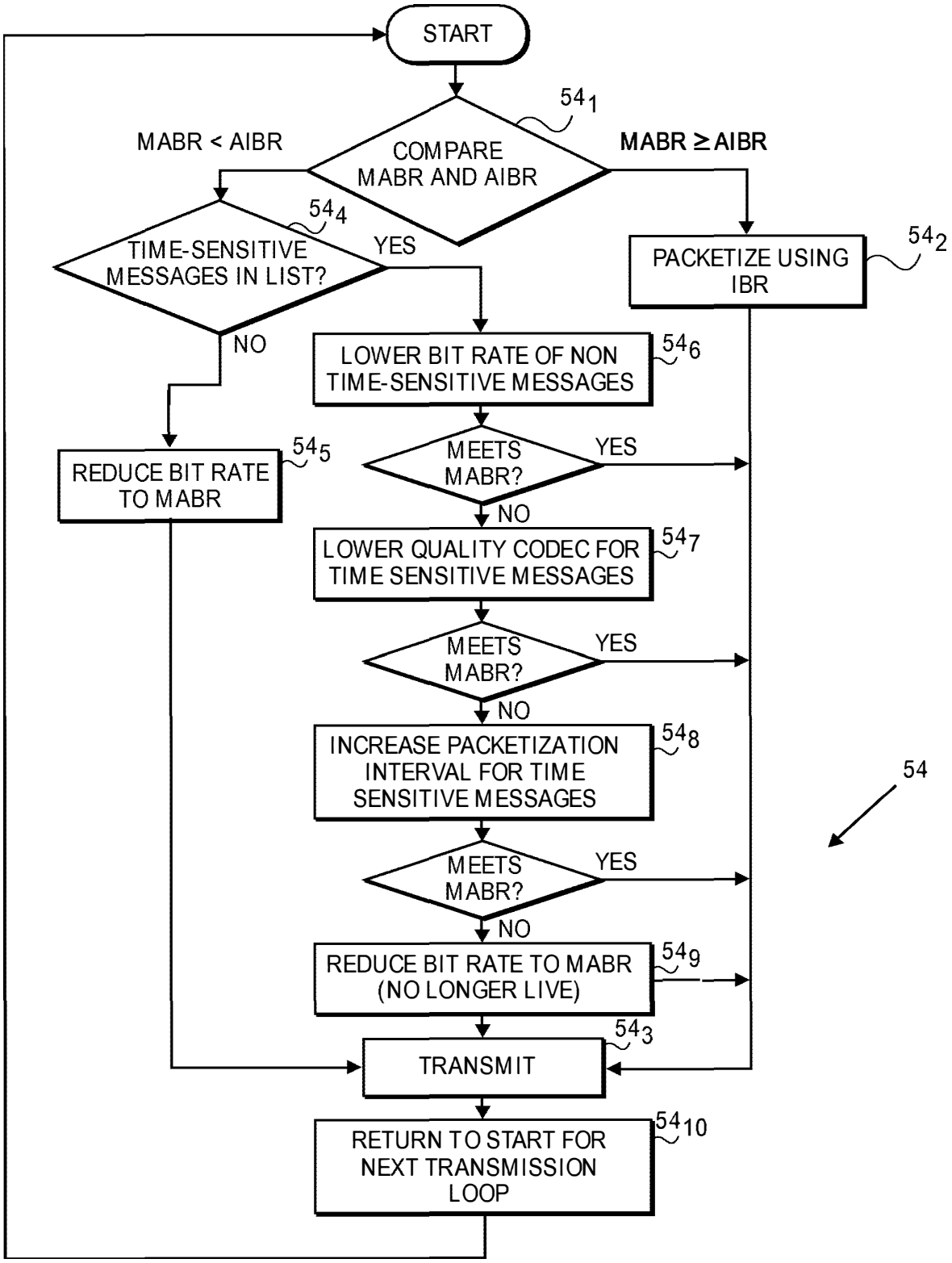

START
COMPARE MABR AND AIBR
$54_1$
MABR < AIBR
MABR ≥ AIBR
TIME-SENSITIVE MESSAGES IN LIST?
$54_4$
YES
NO
PACKETIZE USING IBR
$54_2$
LOWER BIT RATE OF NON TIME-SENSITIVE MESSAGES
$54_6$
REDUCE BIT RATE TO MABR
$54_5$
MEETS MABR?
YES
NO
LOWER QUALITY CODEC FOR TIME SENSITIVE MESSAGES
$54_7$
MEETS MABR?
YES
NO
INCREASE PACKETIZATION INTERVAL FOR TIME SENSITIVE MESSAGES
$54_8$
MEETS MABR?
YES
NO
REDUCE BIT RATE TO MABR (NO LONGER LIVE)
$54_9$
TRANSMIT
$54_3$
RETURN TO START FOR NEXT TRANSMISSION LOOP
$54_{10}$
54

FIG. 9C

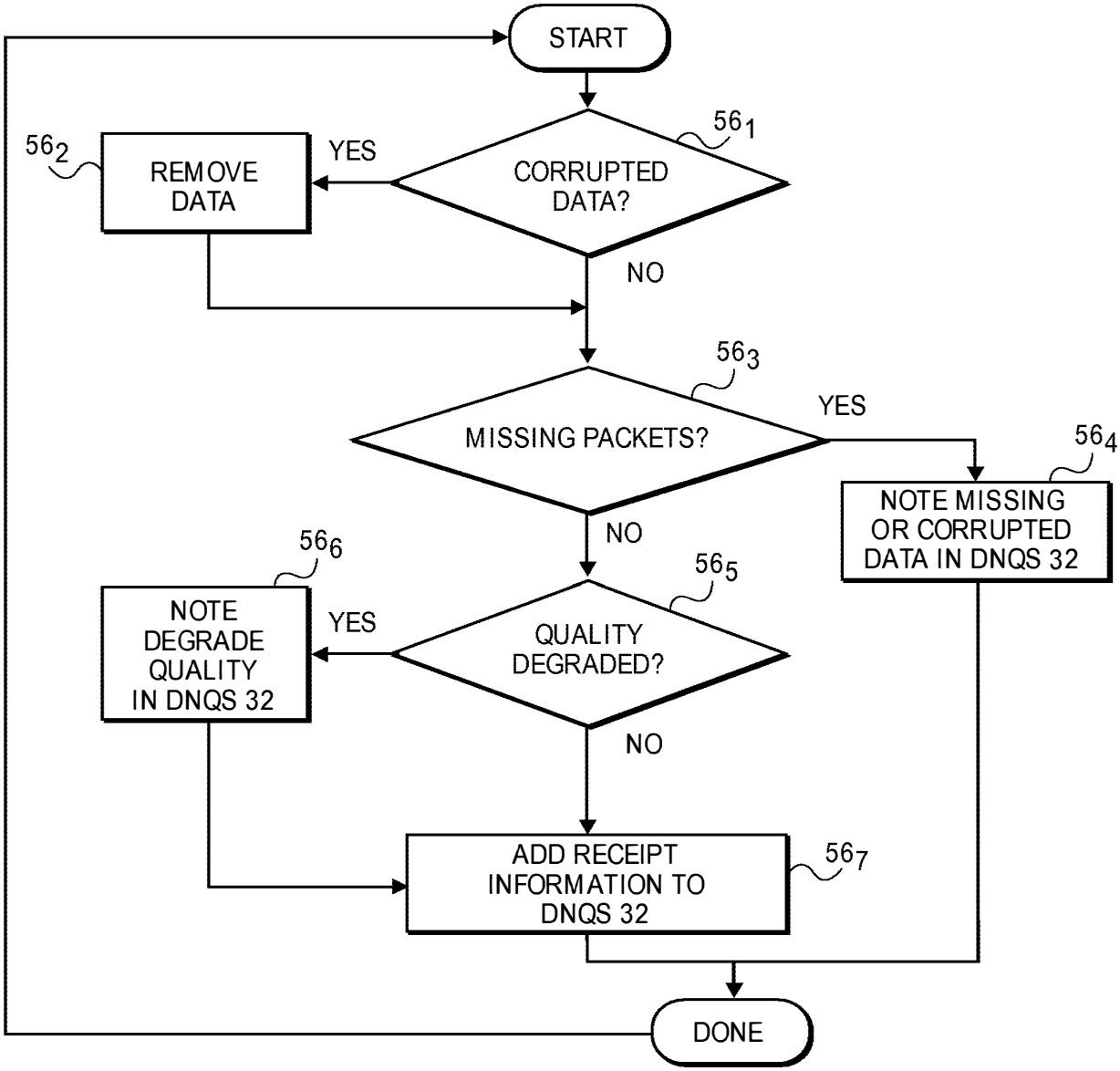

START
561
CORRUPTED DATA?
YES
562
REMOVE DATA
NO
563
MISSING PACKETS?
YES
564
NOTE MISSING OR CORRUPTED DATA IN DNQS 32
NO
565
QUALITY DEGRADED?
YES
566
NOTE DEGRADE QUALITY IN DNQS 32
NO
567
ADD RECEIPT INFORMATION TO DNQS 32
DONE

FIG. 9D

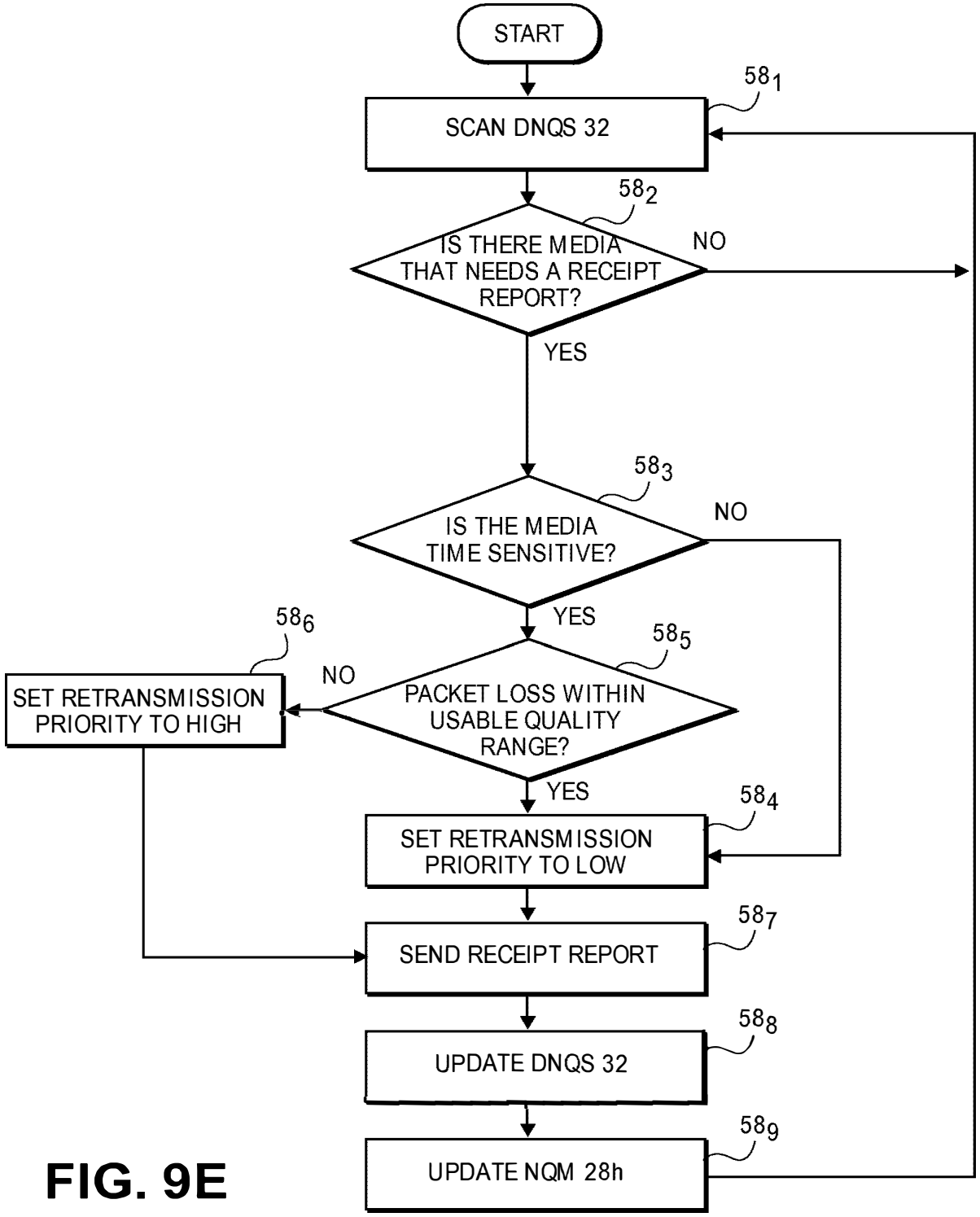

START
SCAN DNQS 32
581
IS THERE MEDIA THAT NEEDS A RECEIPT REPORT?
582
NO
YES
IS THE MEDIA TIME SENSITIVE?
583
NO
YES
PACKET LOSS WITHIN USABLE QUALITY RANGE?
585
NO
SET RETRANSMISSION PRIORITY TO HIGH
586
YES
SET RETRANSMISSION PRIORITY TO LOW
584
SEND RECEIPT REPORT
587
UPDATE DNQS 32
588
UPDATE NQM 28h
589

FIG. 9E

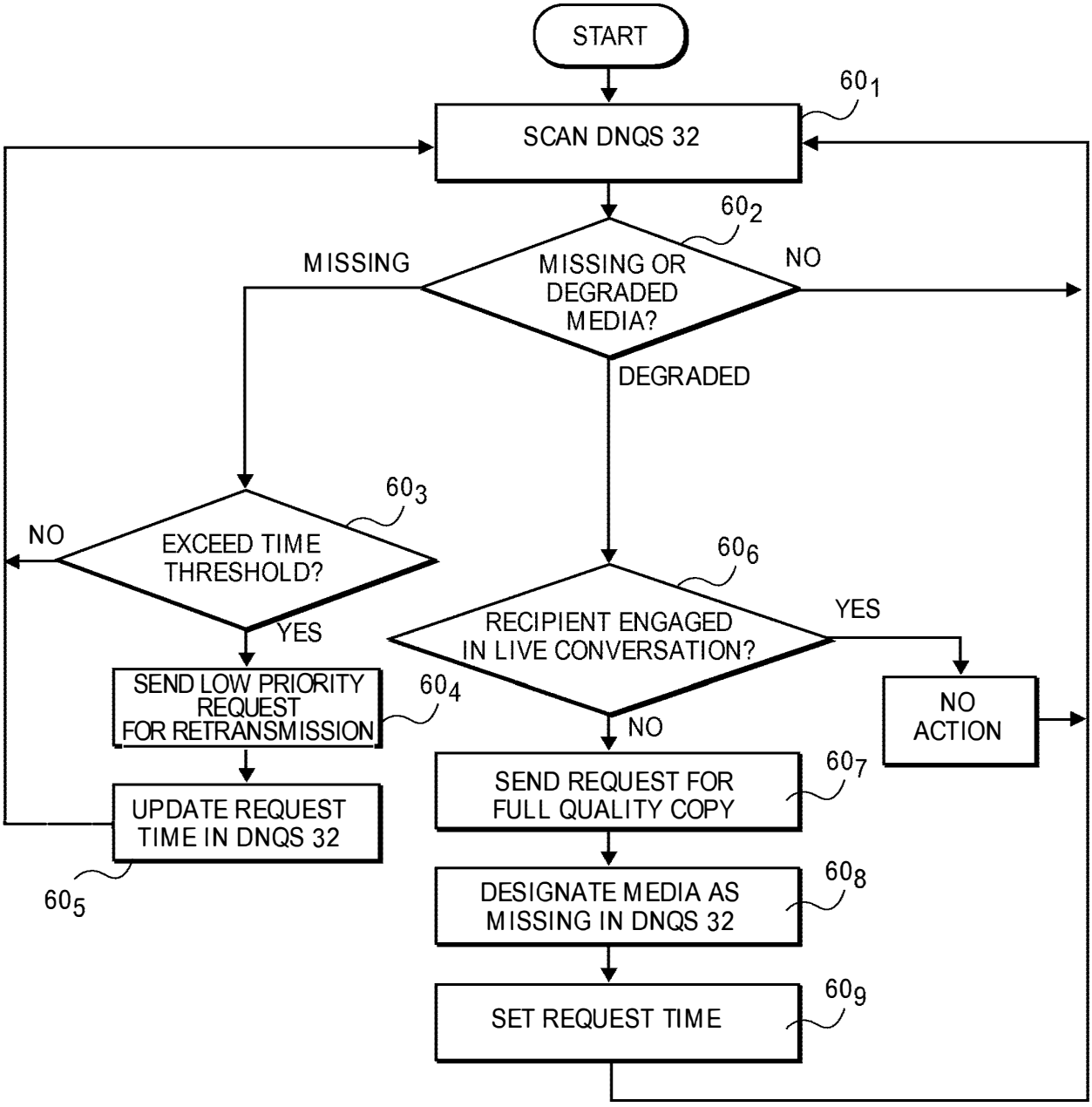

START
SCAN DNQS 32
601
MISSING OR DEGRADED MEDIA?
602
MISSING
NO
DEGRADED
EXCEED TIME THRESHOLD?
603
NO
YES
SEND LOW PRIORITY REQUEST FOR RETRANSMISSION
604
UPDATE REQUEST TIME IN DNQS 32
605
RECIPIENT ENGAGED IN LIVE CONVERSATION?
606
YES
NO
NO ACTION
SEND REQUEST FOR FULL QUALITY COPY
607
DESIGNATE MEDIA AS MISSING IN DNQS 32
608
SET REQUEST TIME
609

FIG. 9F

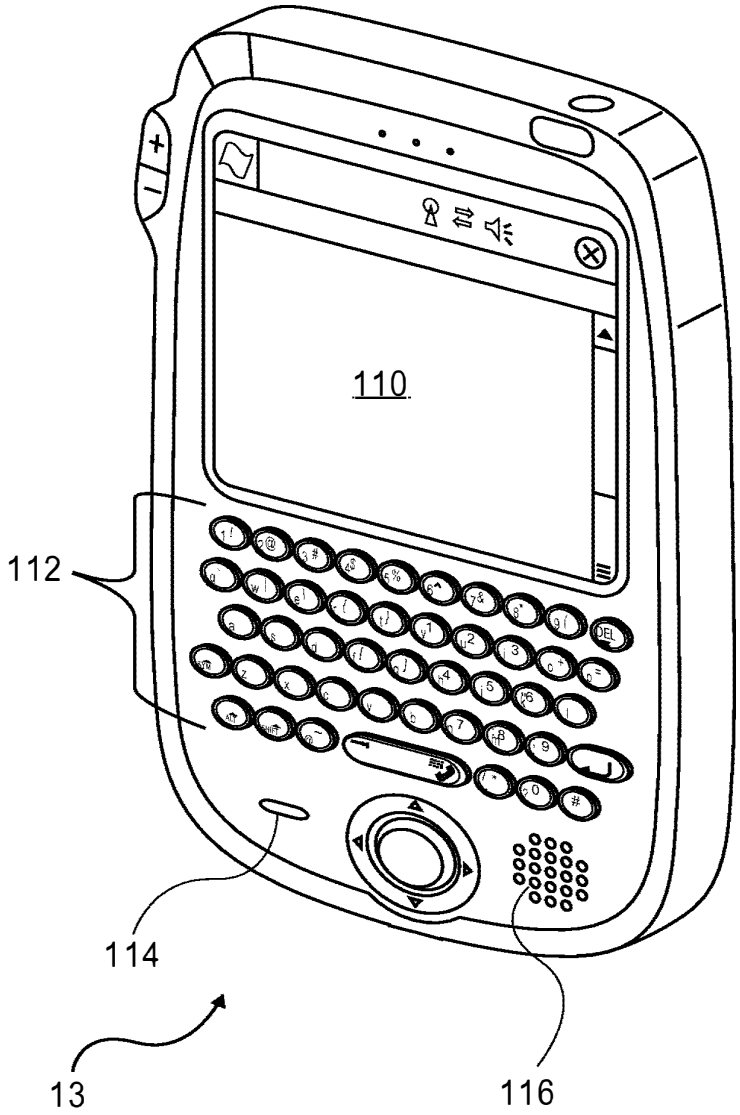



FIG. 10

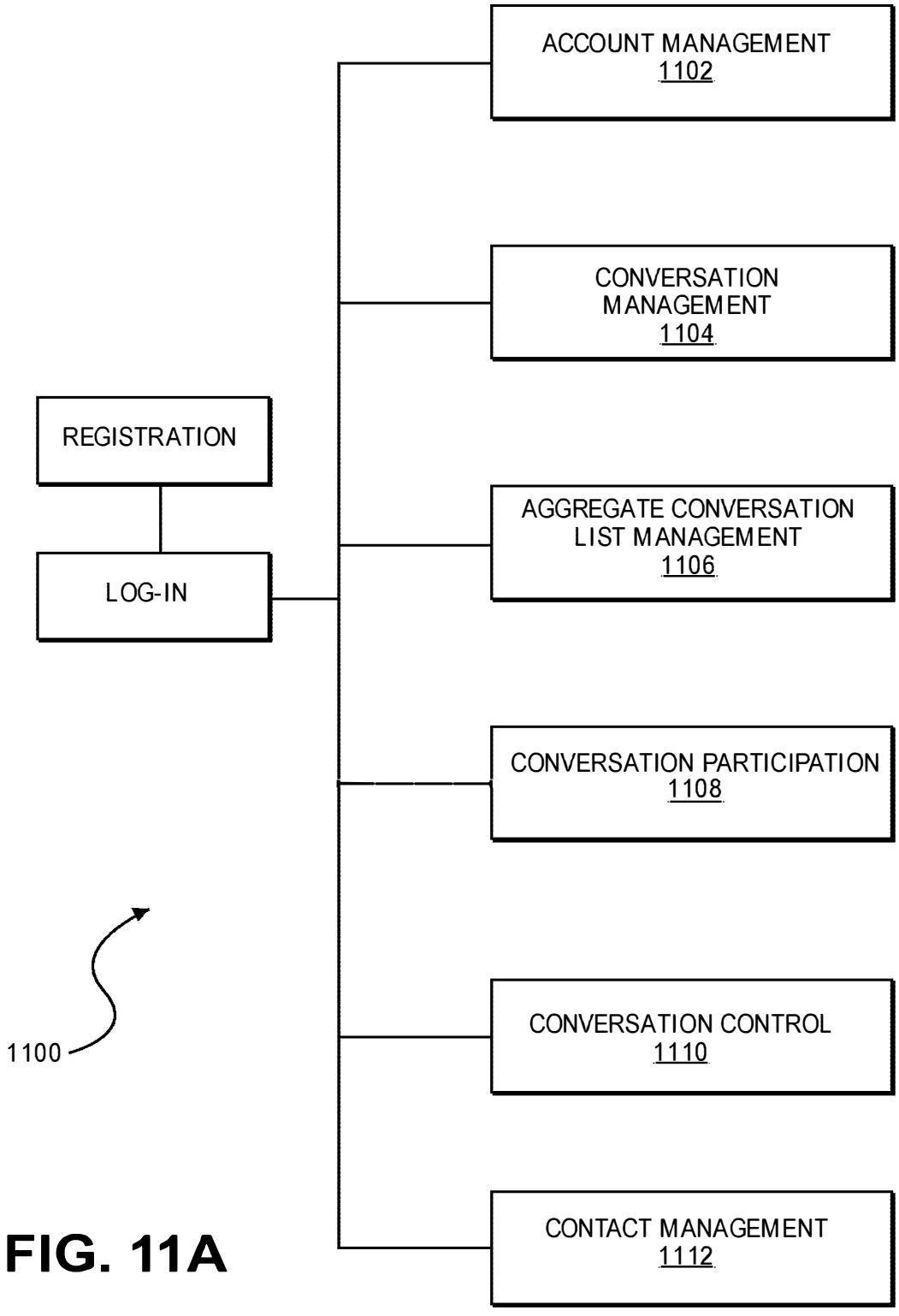

ACCOUNT MANAGEMENT
1102
CONVERSATION MANAGEMENT
1104
REGISTRATION
AGGREGATE CONVERSATION LIST MANAGEMENT
1106
LOG-IN
CONVERSATION PARTICIPATION
1108
1100
CONVERSATION CONTROL
1110
CONTACT MANAGEMENT
1112

FIG. 11A

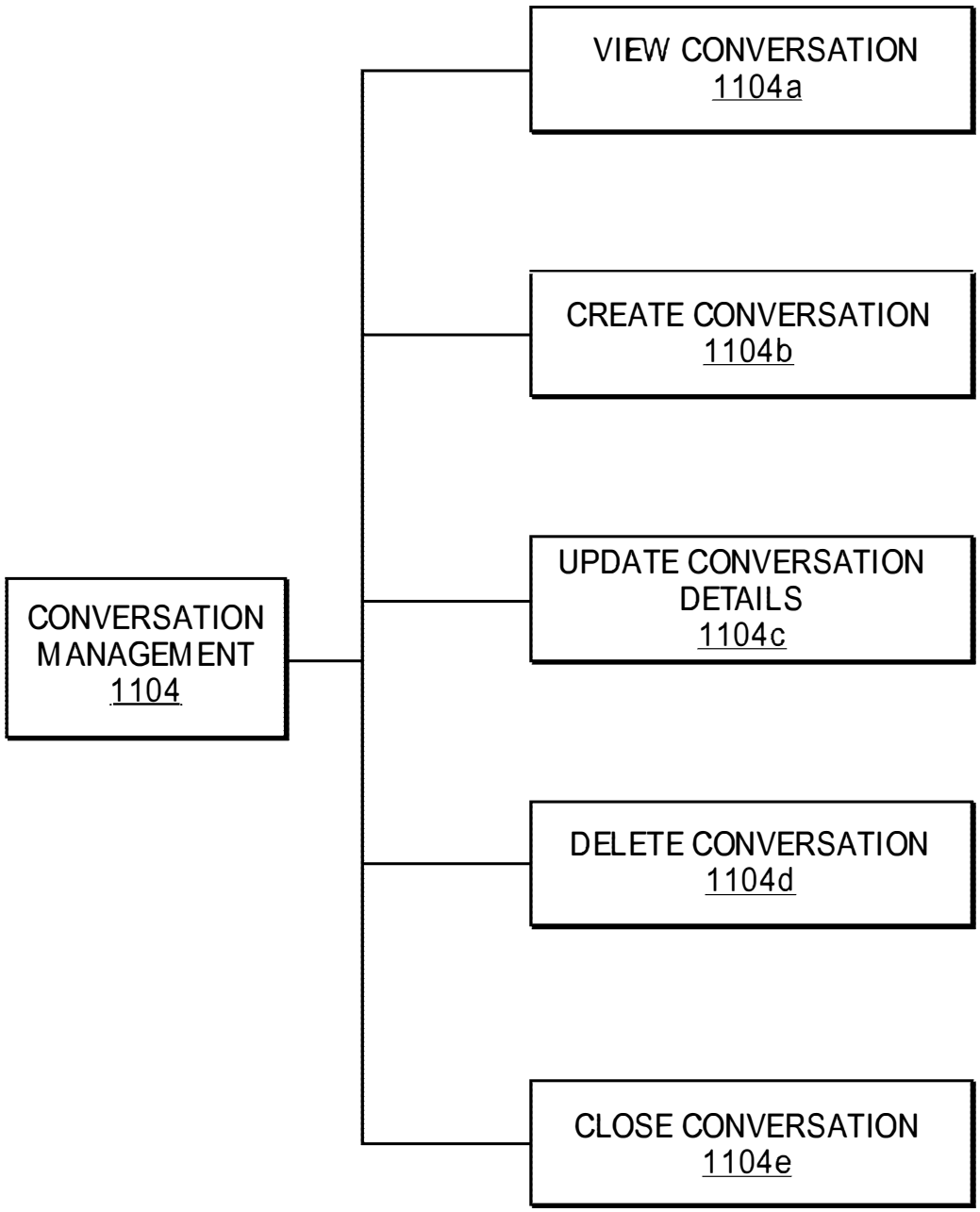

CONVERSATION MANAGEMENT
1104
VIEW CONVERSATION
1104a
CREATE CONVERSATION
1104b
UPDATE CONVERSATION DETAILS
1104c
DELETE CONVERSATION
1104d
CLOSE CONVERSATION
1104e

FIG. 11B

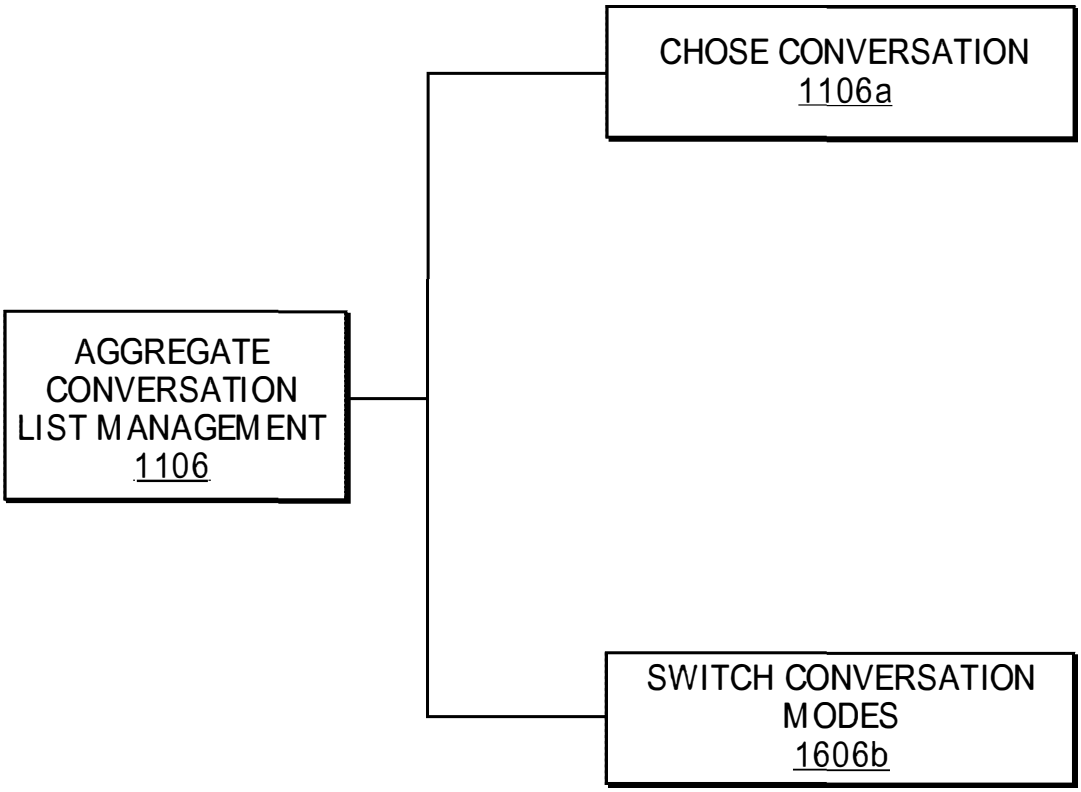

CHOSE CONVERSATION
1106a
AGGREGATE
CONVERSATION
LIST MANAGEMENT
1106
SWITCH CONVERSATION
MODES
1606b

FIG. 11C

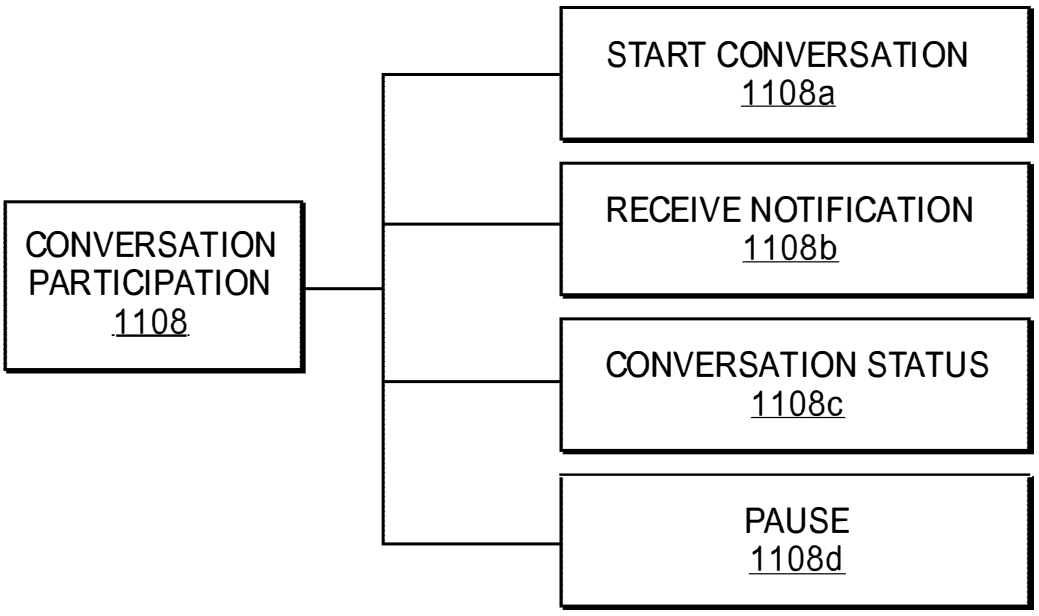



FIG. 11D

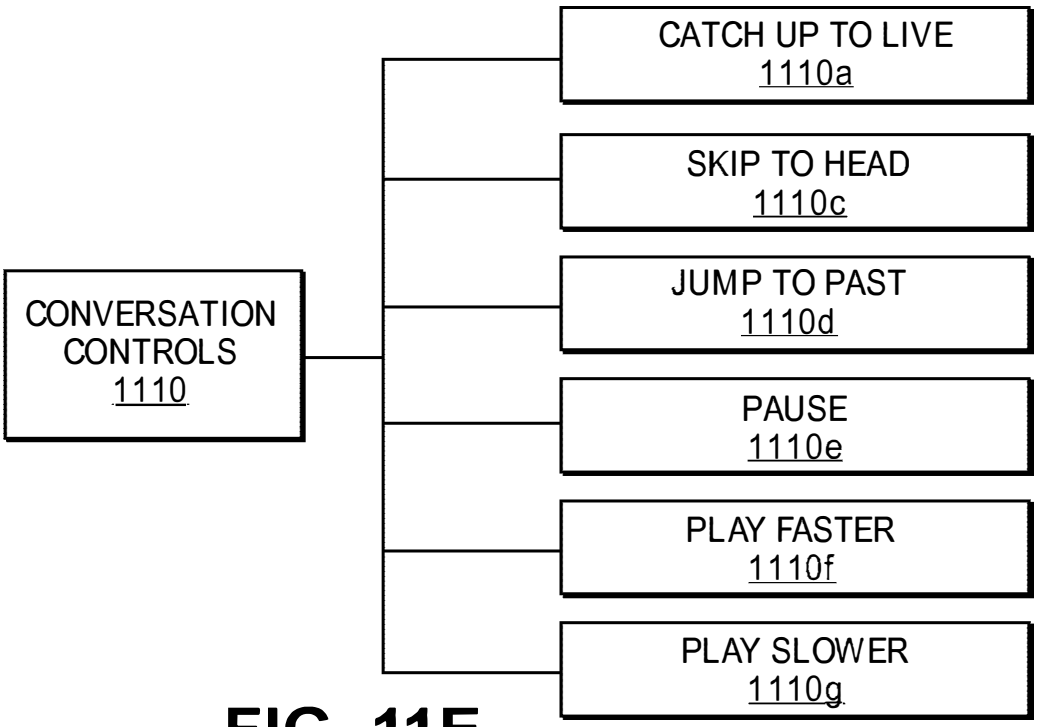



FIG. 11E

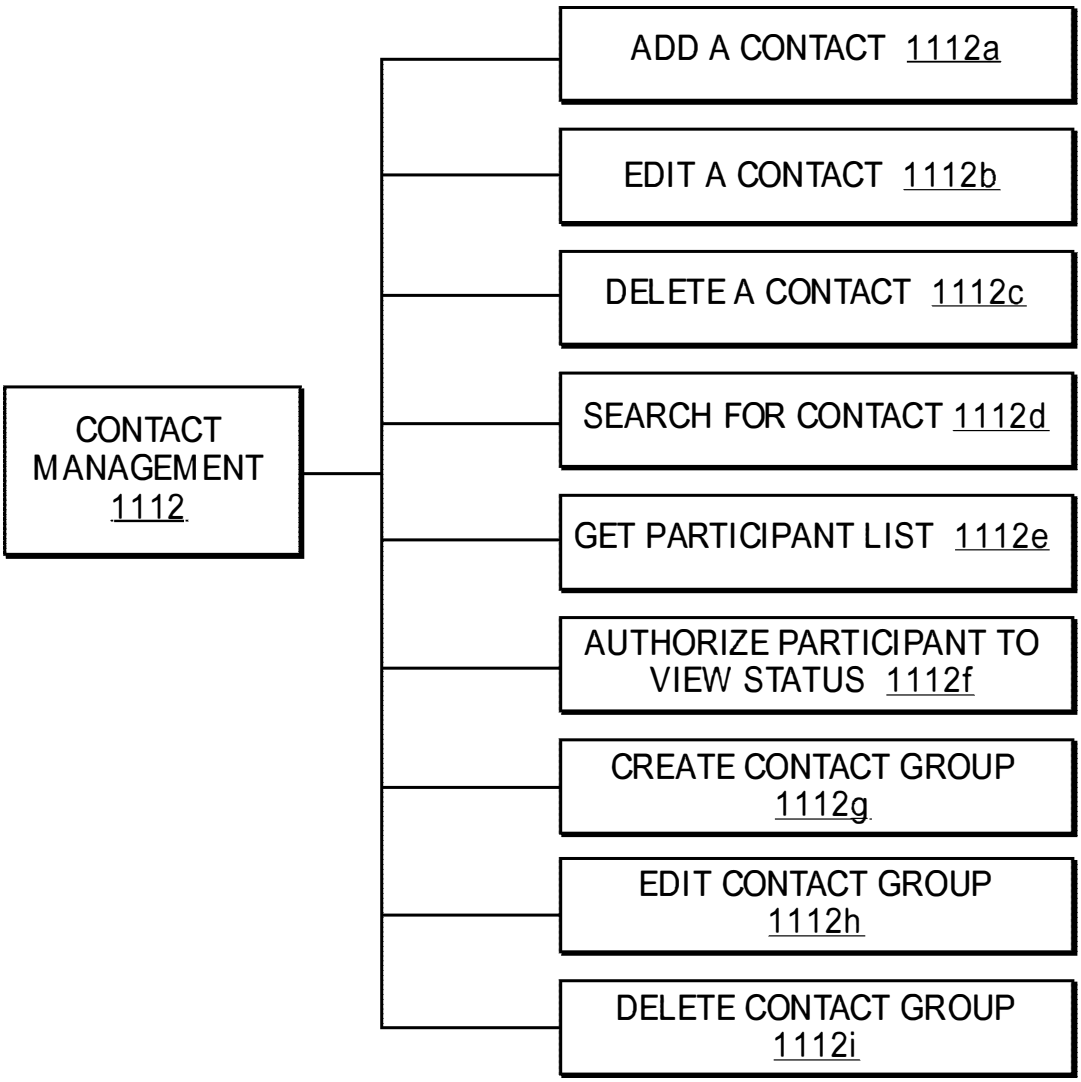

CONTACT MANAGEMENT 1112
ADD A CONTACT 1112a
EDIT A CONTACT 1112b
DELETE A CONTACT 1112c
SEARCH FOR CONTACT 1112d
GET PARTICIPANT LIST 1112e
AUTHORIZE PARTICIPANT TO VIEW STATUS 1112f
CREATE CONTACT GROUP 1112g
EDIT CONTACT GROUP 1112h
DELETE CONTACT GROUP 1112i

FIG. 11F

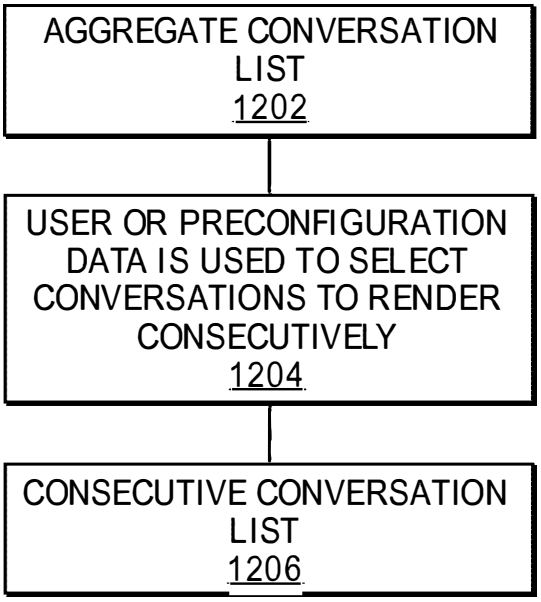



FIG. 12A

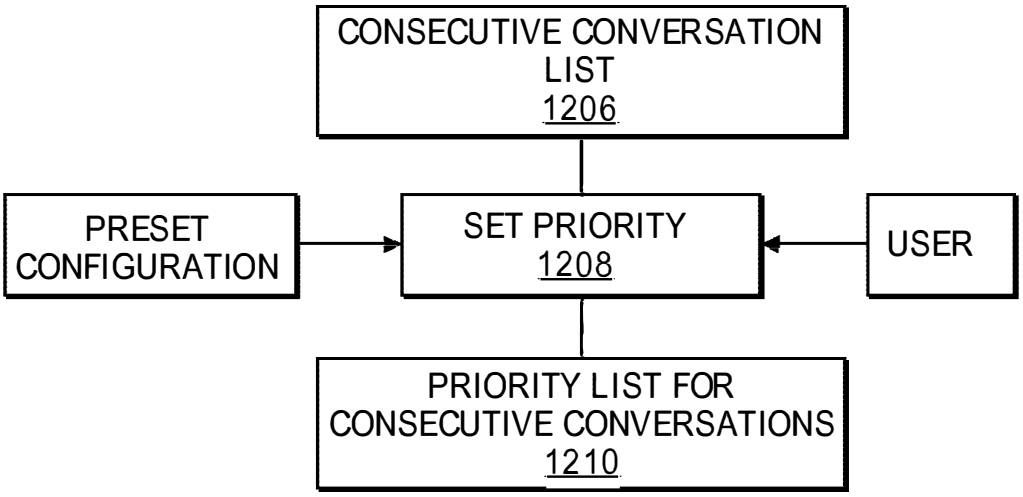



FIG. 12B

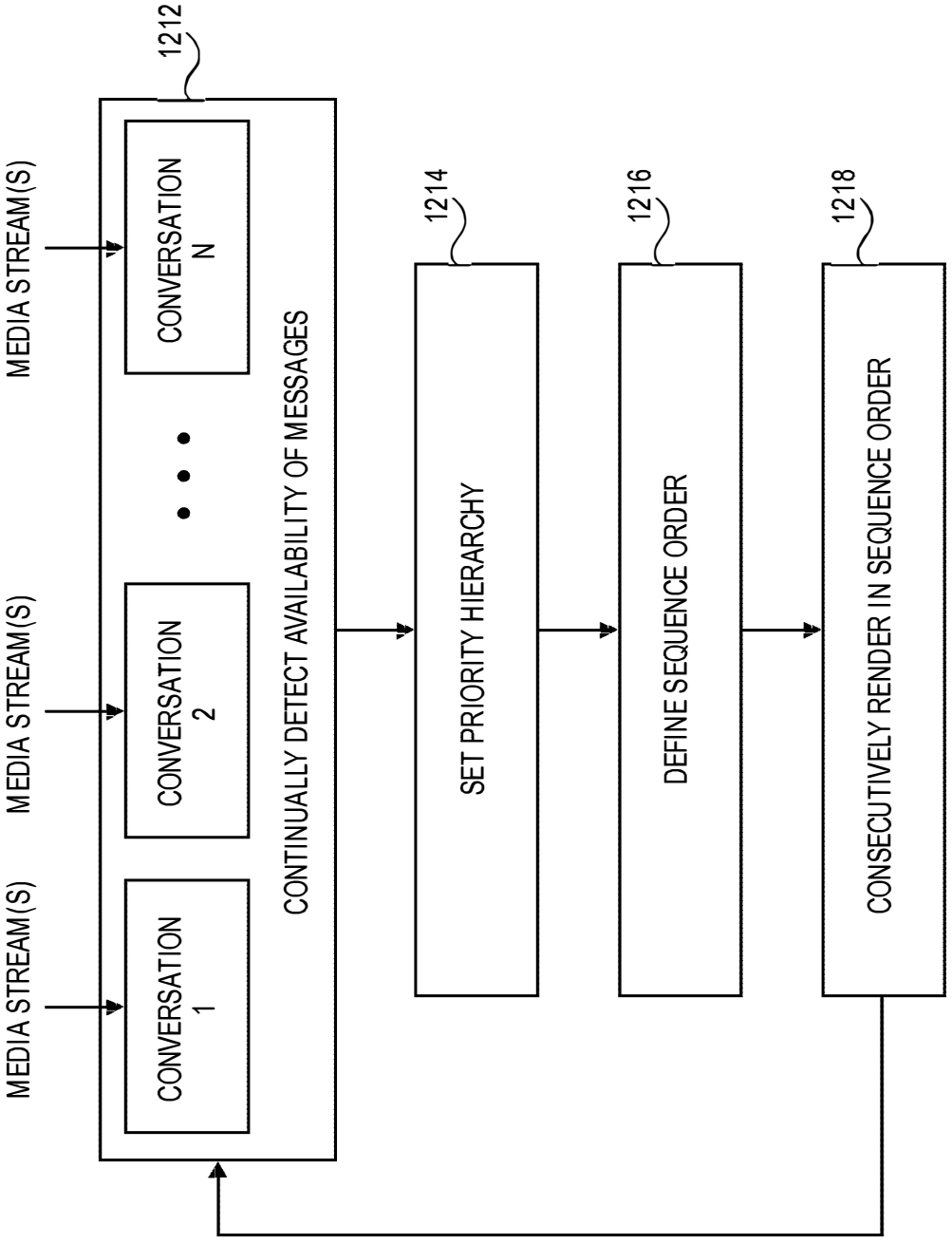

MEDIA STREAM(S)
MEDIA STREAM(S)
MEDIA STREAM(S)
CONVERSATION 1
CONVERSATION 2
CONVERSATION N
CONTINUALLY DETECT AVAILABILITY OF MESSAGES
1212
SET PRIORITY HIERARCHY
1214
DEFINE SEQUENCE ORDER
1216
CONSECUTIVELY RENDER IN SEQUENCE ORDER
1218

FIG. 12C

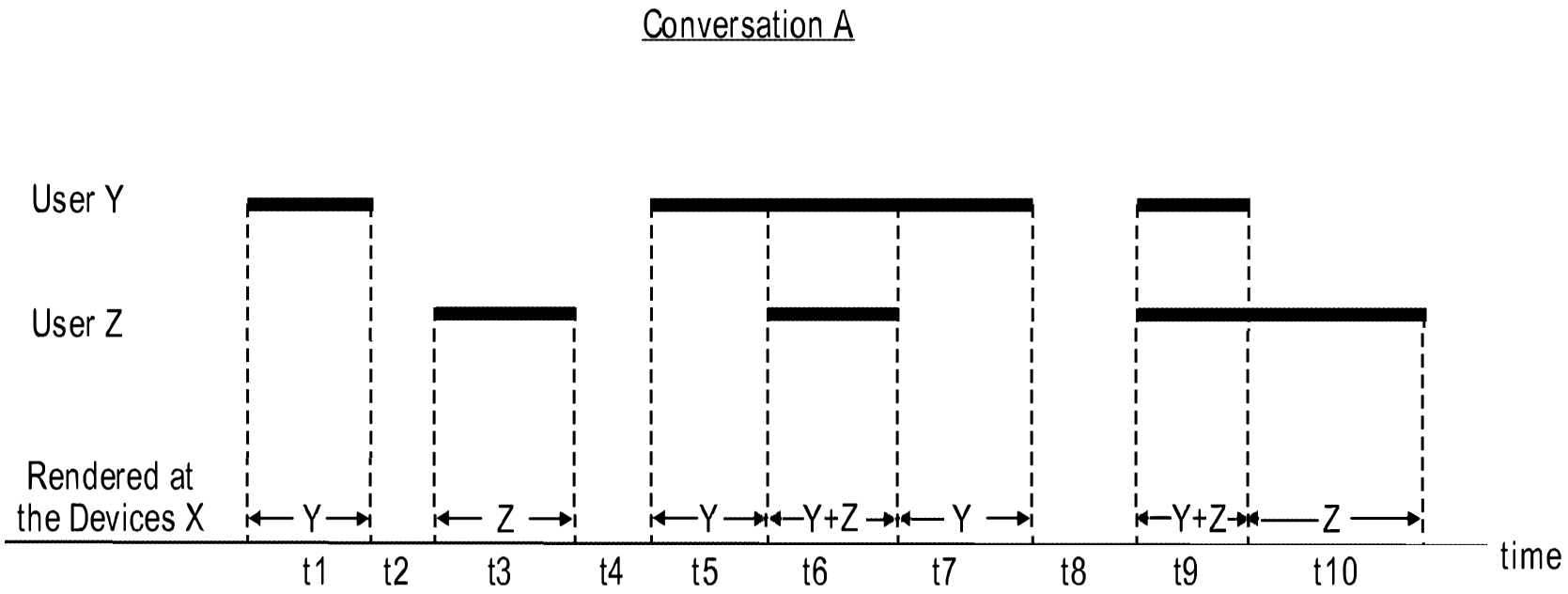
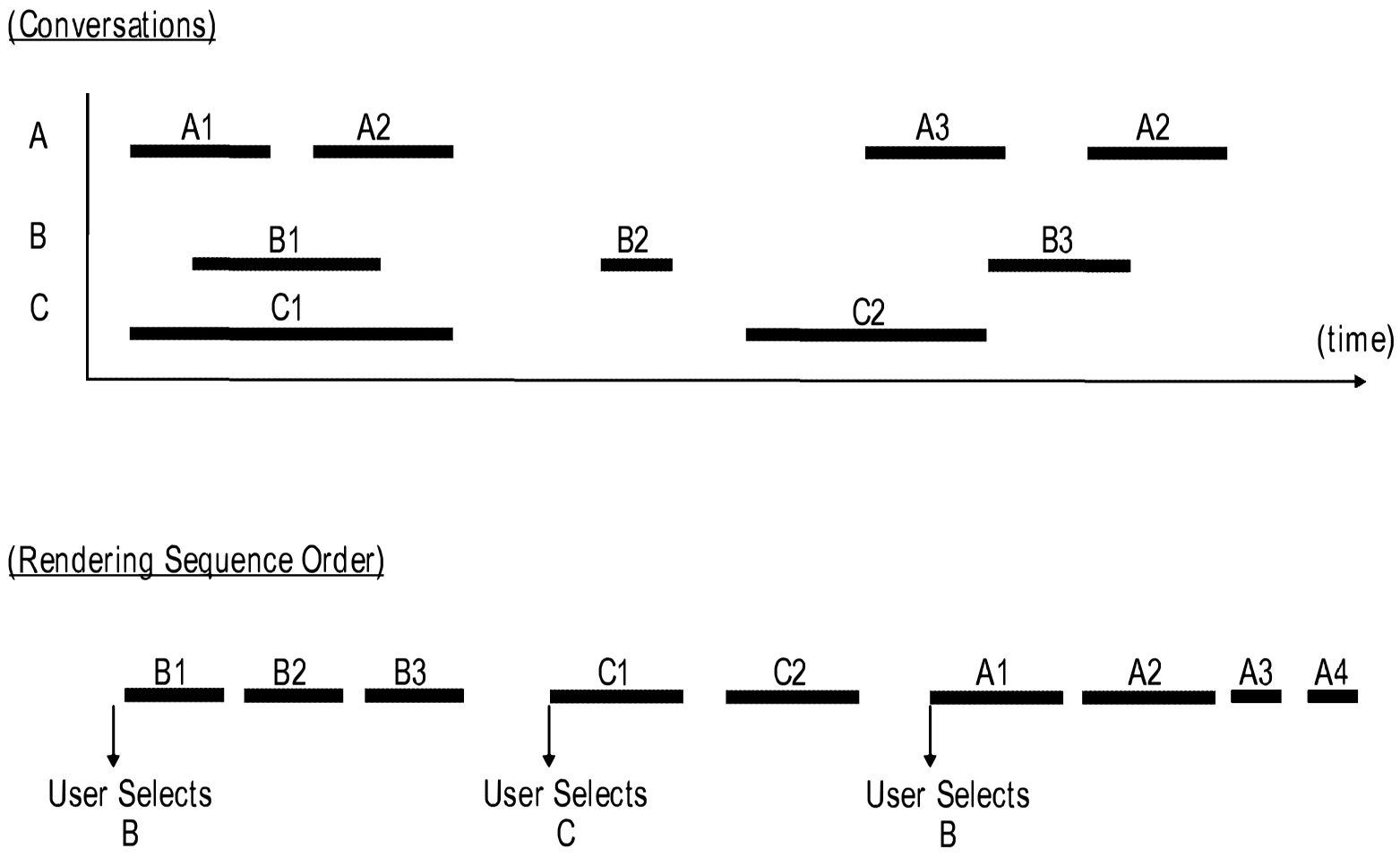
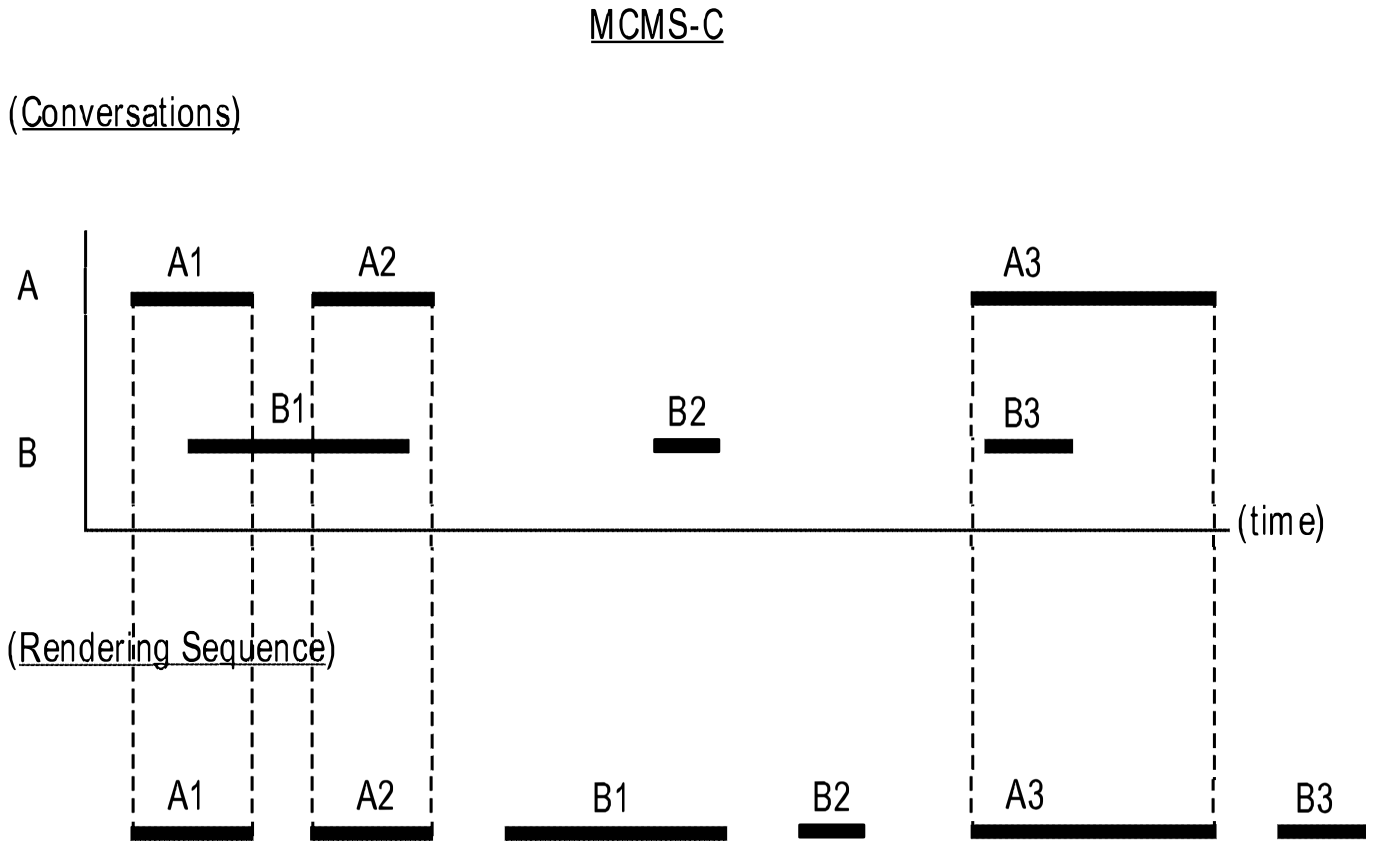

Conversation A
User Y
User Z
Rendered at the Devices X
Y
Z
Y
Y+Z
Y
Y+Z
Z
time
t1
t2
t3
t4
t5
t6
t7
t8
t9
t10
108

FIG. 13A


(Conversations)
A
B
C
A1
A2
A3
A2
B1
B2
B3
C1
C2
(time)
(Rendering Sequence Order)
B1
B2
B3
C1
C2
A1
A2
A3
A4
User Selects B
User Selects C
User Selects B

109

FIG. 13B


MCMS-C
(Conversations)
A
B
A1
A2
A3
B1
B2
B3
(time)
(Rendering Sequence)
A1
A2
B1
B2
A3
B3

110

FIG. 13C

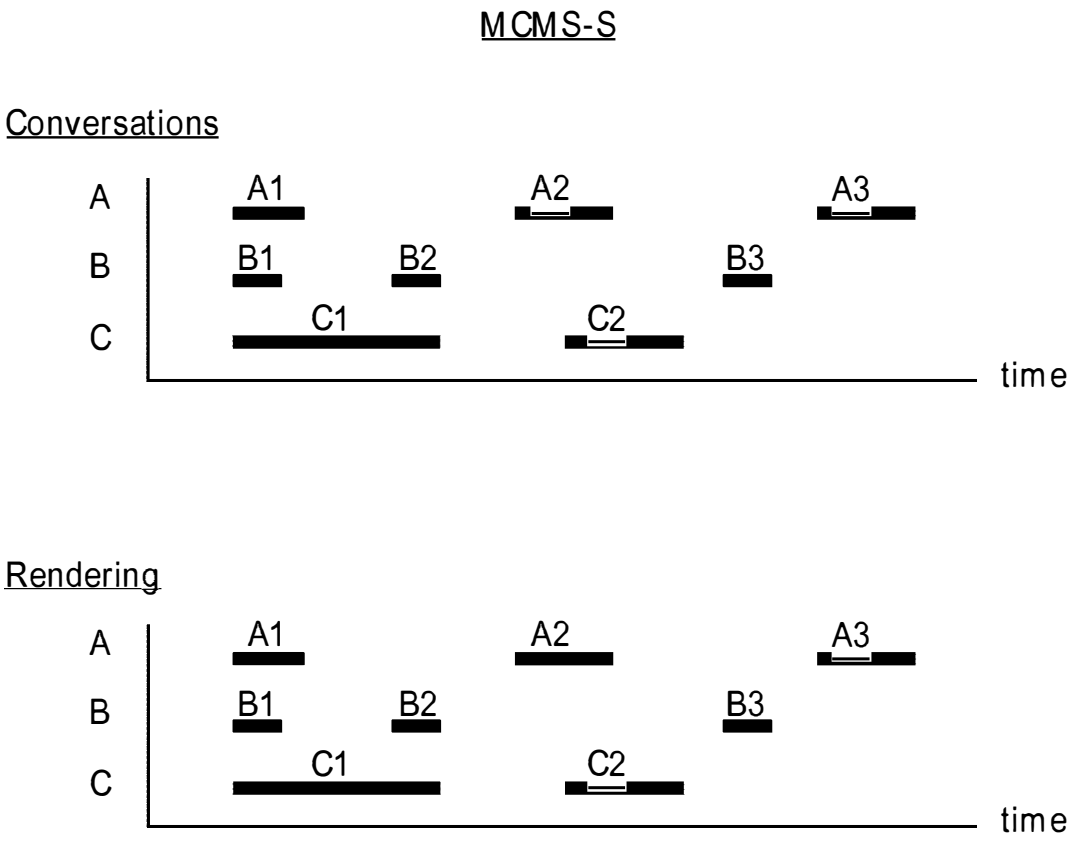

MCMS-S
Conversations
A
B
C
A1
A2
A3
B1
B2
B3
C1
C2
time
Rendering
A
B
C
A1
A2
A3
B1
B2
B3
C1
C2
time

FIG. 13D

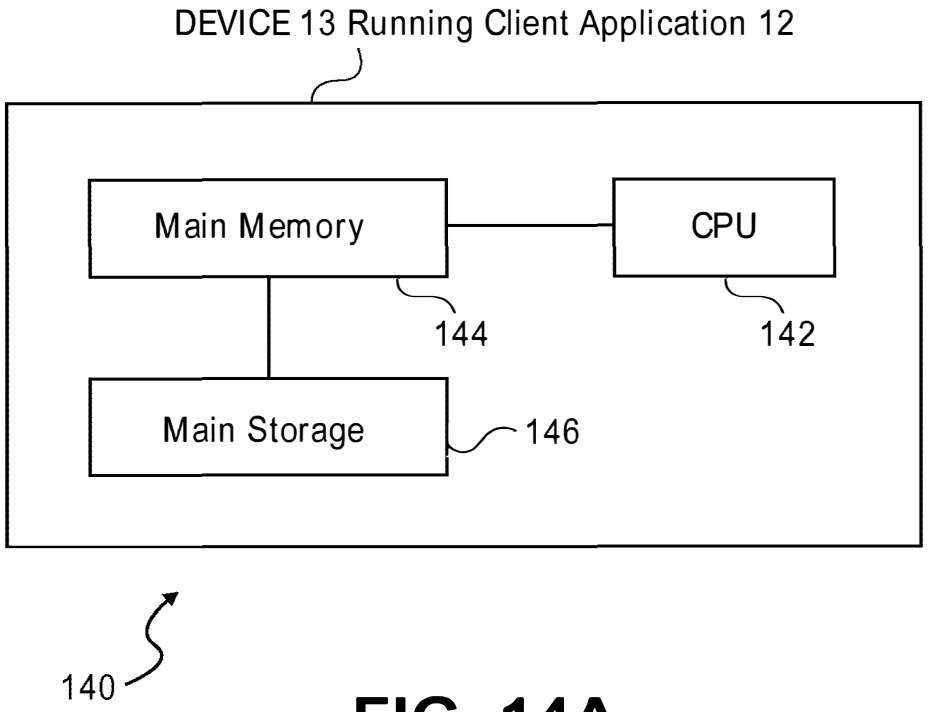



FIG. 14A

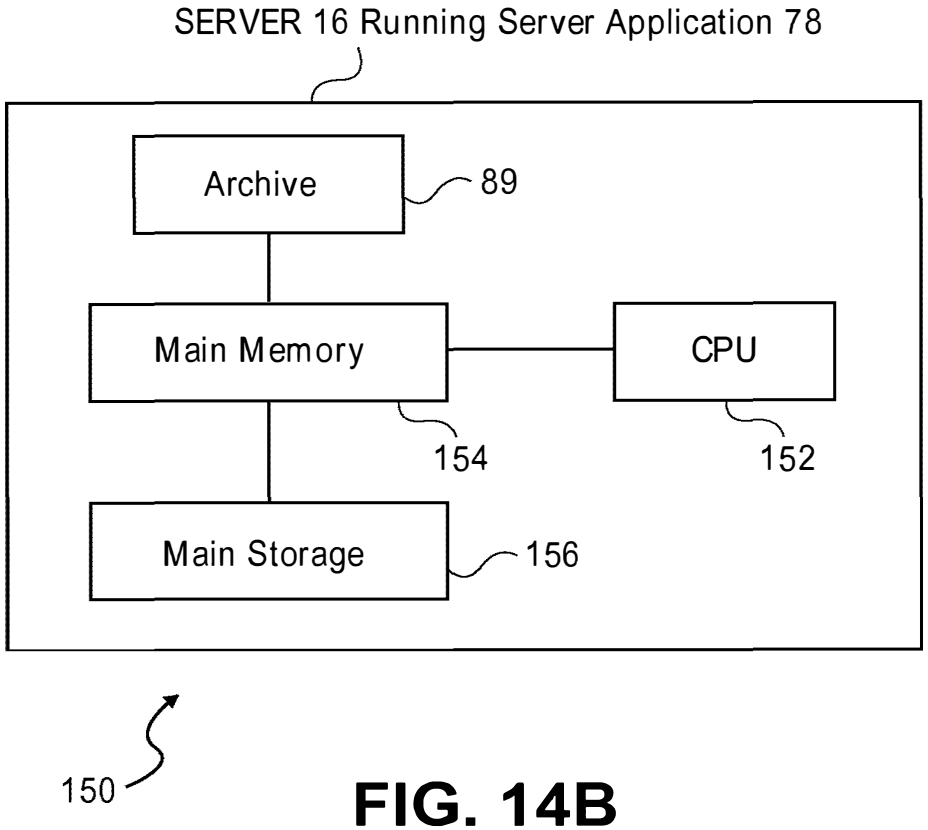



FIG. 14B

# Electronic Patent Application Fee Transmittal

| | |
|---|---|
| **Application Number:** | |
| **Filing Date:** | |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| First Named Inventor/Applicant Name: | Thomas E. KATIS |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |

Filed as Small Entity

**Utility Filing Fees**

| Description | Fee Code | Quantity | Amount | Sub-Total in USD($) |
|---|---|---|---|---|
| **Basic Filing:** | | | | |
| Utility filing Fee (Electronic filing) | 4011 | 1 | 75 | 75 |
| Utility Search Fee | 2111 | 1 | 255 | 255 |
| Utility Examination Fee | 2311 | 1 | 105 | 105 |
| **Pages:** | | | | |
| **Claims:** | | | | |
| Claims in excess of 20 | 2202 | 2 | 25 | 50 |
| **Miscellaneous-Filing:** | | | | |
| **Petition:** | | | | |

| Description | Fee Code | Quantity | Amount | Sub-Total in USD($) |
|---|---|---|---|---|
| **Patent-Appeals-and-Interference:** | | | | |
| Post-Allowance-and-Post-Issuance: | | | | |
| **Extension-of-Time:** | | | | |
| **Miscellaneous:** | | | | |
| | | **Total in USD ($)** | | **485** |

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 2907698 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. KATIS |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 26-FEB-2008 |
| **Filing Date:** | |
| **Time Stamp:** | 13:00:07 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | yes |
| Payment Type | Deposit Account |
| Payment was successfully received in RAM | $485 |
| RAM confirmation Number | 8252 |
| Deposit Account | 500388 |
| Authorized User | |

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

Charge any Additional Fees required under 37 C.F.R. Section 1.16 (National application filing, search, and examination fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes) /Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Transmittal of New Application | Transmittal_New_Application.pdf | 32794<br>334d4b49fa613**8**8f3cd2ab3930c5d663a6c6015c | no | 3 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 2 | Application Data Sheet | APPLICATION_DATA_SHEET.pdf | 1159356<br>3**8**d396b9a1ca40e1bfb6d7aa9c02c7336c0a2f30 | no | 5 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 3 | Oath or Declaration filed | Declaration.pdf | 754254<br>9c7b4**8**29404ac371a609039a7b39610**88**4bd4279 | no | 2 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 4 | | REBLP003_as_filed.pdf | 318704<br>42425b1f0a36ab242**8**70**88**42c7f3d**8**0bd0946954 | yes | 71 |

**Multipart Description/PDF files in .zip description**

| Document Description | Start | End |
|---|---|---|
| Specification | 1 | 66 |
| Claims | 67 | 70 |
| Abstract | 71 | 71 |

| | | | | | |
|---|---|---|---|---|---|
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 5 | Drawings-only black and white line drawings | REBLP003_Figures.pdf | 275020<br>ecf4**8**ba051194f162010006ae95cbce1**8**a3a929e | no | 33 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 6 | Fee Worksheet (PTO-06) | fee-info.pdf | 8518<br>f**8**049ad**8**de**8**54bf6496742e2fb49fd1a4a62a64e | no | 2 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| | | **Total Files Size (in bytes):** | 2548646 | | |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**

**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**

**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**

**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on February 26, 2008.

Signed: /Laura M. Dean/
Laura M. Dean

Attorney Docket No.:
REBLP003/VOX-P003
First Named Inventor: KATIS et al.

**UTILITY PATENT APPLICATION TRANSMITTAL (37 CFR. § 1.53(b))**
(Continuation, Divisional or Continuation-in-part application)

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir: This is a request for filing a patent application under 37 CFR. § 1.53(b) in the name of inventors:
**Thomas E. KATIS, James T. PANTTAJA, Mary G. PANTTAJA and Matthew J. RANNEY**

For: **TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS**

Assigned to : RebelVox, LLC
This application is a ☒ Continuation ☐ Divisional ☐ Continuation-in-part of prior Application No.: 12/028,400, from which priority under 35 U.S.C. §120 is claimed. Prior Application No.: 12/028,400 claims the benefit of priority under 35 U.S.C. § 119(e) to U.S. provisional application No. 60/937,552 filed on June 28, 2007, and U.S. provisional application No. 60/999,619 filed on October 19, 2007.

The specification has been amended to claim priority from the parent application, or such amendment is included in a separate sheet.

Application Elements:

☒ **71** Pages of Specification, Claims and Abstract
☒ **33** Sheets of formal Drawings
☒ Declaration and Power of Attorney
- ☒ Newly executed
- ☐ Copy from a prior application (37 CFR 1.63(d) for a continuation or divisional). The entire disclosure of the prior application from which a copy of the declaration is

herein supplied is considered as being part of the disclosure of the accompanying application and is hereby incorporated by reference therein.

☐ Deletion of inventors. Signed statement attached deleting inventor(s) named in the prior application, see 37 CFR 1.63(d)(2) and 1.33(b).

Accompanying Application Parts:

☐ Do not publish this application. Nonpublication Request is attached.

☐ Assignment and Assignment Recordation Cover Sheet (recording fee of $40.00 enclosed)

☐ Power of Attorney

☐ 37 CFR 3.73(b) Statement by Assignee

☐ Information Disclosure Statement with Form PTO-1449 ☐ Copies of IDS Citations

☐ Preliminary Amendment (*New claims numbered after highest original claim in prior application.*)

☐ Return Receipt Postcard

☒ Other: Application Data Sheet

Extension of Time for Prior Pending Application

☐ A Petition for Extension of Time is being concurrently filed in the prior pending application. A copy of the Petition for Extension of Time is attached.

Fee Calculation (37 CFR § 1.16)

☒ Applicant is entitled to Small Entity Status under 37 C.F.R. §1.27.

| | (Col. 1) Total Claims | | (Col. 2) Claims | (Col. 3) Present Extra | Rate | Additional Fee |
|---|---|---|---|---|---|---|
| TOTAL | 22 | MINUS | 20 | = 3 | x 50 = | 100 |
| INDEP. | 1 | MINUS | 3 | = 0 | x 210 = | |
| | | | | Utility Search Fee | = | $510 |
| | | | | Utility Examination Fee | = | $210 |
| | | | | Utility Application Size Fee (for each additional 50 sheets that exceeds 100 sheets) | $260 | |
| | | | | [ ] First presentation of multiple dependent claim | $370 | |
| | | | | Basic Filing Fee under 37 C.F.R. §1.16(a) | = | $310 |
| | | | | | TOTAL | 1,130.00 |
| | | | | | SMALL ENTITY 50% FILING FEE REDUCTION (if applicable) | $565.00 |
| | | | | | Further SMALL ENTITY Filing Fee Reduction for Electronic Filing (- $80.00) | $485.00 |

☒ The Commissioner is authorized to charge any fees beyond the amount enclosed which may be required, or to credit any overpayment, to Deposit Account No. 500388 (Order No. REBLP003).

General Authorization for Petition for Extension of Time (37 CFR §1.136)

☒ Applicants hereby make and generally authorize any Petitions for Extensions of Time as may be needed for any subsequent filings. The Commissioner is also authorized to charge any extension fees under 37 CFR §1.17 as may be needed to Deposit Account No. 500388 (Order No. REBLP003).

☒ Please send correspondence to the following address:

**Customer Number 76743**

Date: February 26, 2008

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

# DECLARATION AND POWER OF ATTORNEY FOR ORIGINAL U.S. PATENT APPLICATION

As a below-named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe that I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled: TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS the specification of which,

(check one)

1. ☒ is attached hereto.

2. ☐ was filed on _____ as
U.S. Application No. _____
and was amended on _____.

3. ☐ was filed on _____ as
PCT International Application No. _____
and was amended on _____.

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to patentability as defined in 37 CFR 1.56, including for continuation-in-part applications, material information which became available between the filing date of the prior application and the national or PCT international filing date of the continuation-in-part application.

**Prior Foreign Application(s)**

I hereby claim foreign priority benefits under Title 35, United States Code, §119(a)-(d) or (f), or §365(b) of any foreign application for patent or inventor's certificate listed below, or under §365(a) of any PCT international application which designated at least one country other than the United States listed below, and I have identified below, by checking the box, any foreign application for patent or inventor's certificate, or any PCT International application having a filing date before that of the application on which priority is claimed:

| | | | Priority Not Claimed | Cert. Copy Attached |
|---|---|---|---|---|
| _____ (Application No.) | _____ (Country) | _____ (Filing Date) | ☐ | ☐ |
| _____ (Application No.) | _____ (Country) | _____ (Filing Date) | ☐ | ☐ |

**Power of Attorney**

And I hereby appoint all practitioners who are associated with the Customer Number 76743 as my principal attorneys to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith.

**Direct Correspondence To:** **Customer Number 76743**

**Direct Telephone Calls To:** **Steve D Beyer at telephone number (408) 255-8001**

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further 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, and that such willful false statements may jeopardize the validity of the application or any patent issuing thereon.

Typewritten Full Name of Sole or First Inventor: Thomas E. Katis Citizenship: US

**Inventor's signature:** [signature] **Date of Signature:** 2/21/08

Residence: (City) Jackson (State/Country) WY/US

Post Office Address: 970 East Broadway, E-321, Jackson, WY 83001-9475

Second Inventor: James T. Panttaja Citizenship: US

**Inventor's signature:** [signature] **Date of Signature:** 2-21-08

Residence: (City) Healdsburg (State/Country) CA/US

Post Office Address: 4016 West Soda Rock Lane, Healdsburg, CA 95448

Third Inventor: Mary G. Panttaja Citizenship: US

**Inventor's signature:** [signature] **Date of Signature:** 2-21-08

Residence: (City) Healdsburg (State/Country) CA/US

Post Office Address: 4016 West Soda Rock Lane, Healdsburg, CA 95448

Fourth Inventor: Matthew J. Rainey Citizenship: US

**Inventor's signature:** [signature] **Date of Signature:** 2-21-2008

Residence: (City) Oakland (State/Country) CA/US

Post Office Address: 280 Euclid Avenue, Oakland, CA 94610

PTO/SB/06 (12-04)

**Filing Date: 02/26/08**

Approved for use through 7/31/2006. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

## PATENT APPLICATION FEE DETERMINATION RECORD

Substitute for Form PTO-875

Application or Docket Number: 12/037,256

### APPLICATION AS FILED – PART I

| FOR | (Column 1) NUMBER FILED | (Column 2) NUMBER EXTRA | RATE ($) | FEE ($) | OR | SMALL ENTITY RATE ($) | SMALL ENTITY FEE ($) |
|---|---|---|---|---|---|---|---|
| BASIC FEE (37 CFR 1.16(a), (b), or (c)) | N/A | N/A | N/A | 75 | | N/A | |
| SEARCH FEE (37 CFR 1.16(k), (i), or (m)) | N/A | N/A | N/A | 255 | | N/A | |
| EXAMINATION FEE (37 CFR 1.16(o), (p), or (q)) | N/A | N/A | N/A | 105 | | N/A | |
| TOTAL CLAIMS (37 CFR 1.16(i)) | 22 minus 20 = | 2 | X$ 25 | 50 | OR | X$50 | |
| INDEPENDENT CLAIMS (37 CFR 1.16(h)) | 1 minus 3 = | * 0 | X$105 | 0 | | X$210 | |
| APPLICATION SIZE FEE (37 CFR 1.16(s)) | 11952176 | | | | | | |
| MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j)) | | | 185 | | | 370 | |
| | | | TOTAL | 485 | | TOTAL | 0 |

* If the difference in column 1 is less than zero, enter "0" in column 2.

### APPLICATION AS AMENDED – PART II

| | | (Column 1) CLAIMS REMAINING AFTER AMENDMENT | | (Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR | (Column 3) PRESENT EXTRA | SMALL ENTITY RATE ($) | SMALL ENTITY ADDITIONAL FEE ($) | OR | OTHER THAN SMALL ENTITY RATE ($) | OTHER THAN SMALL ENTITY ADDITIONAL FEE ($) |
|---|---|---|---|---|---|---|---|---|---|---|
| AMENDMENT A | Total (37 CFR 1.16(i)) | * | Minus | ** | = | x = | | OR | x = | |
| | Independent (37 CFR 1.16(h)) | * | Minus | *** | = | x = | | OR | x = | |
| | Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | N/A | | OR | N/A | |
| | | | | | | TOTAL ADD'T FEE | | OR | TOTAL ADD'T FEE | |

| | | (Column 1) CLAIMS REMAINING AFTER AMENDMENT | | (Column 2) HIGHEST NUMBER PREVIOUSLY PAID FOR | (Column 3) PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | OR | RATE ($) | ADDITIONAL FEE ($) |
|---|---|---|---|---|---|---|---|---|---|---|
| AMENDMENT B | Total (37 CFR 1.16(i)) | * | Minus | ** | = | x = | | OR | x = | |
| | Independent (37 CFR 1.16(h)) | * | Minus | *** | = | x = | | OR | x = | |
| | Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | N/A | | OR | N/A | |
| | | | | | | TOTAL ADD'T FEE | | OR | TOTAL ADD'T FEE | |

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.

** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".

*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

*If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.*

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

| APPLICATION NUMBER | FILING or 371(c) DATE | GRP ART UNIT | FIL FEE REC'D | ATTY.DOCKET.NO | TOT CLAIMS | IND CLAIMS |
|---|---|---|---|---|---|---|
| 12/037,256 | 02/26/2008 | 2614 | 485 | REBLP003/VOX-P003 | 22 | 1 |

**CONFIRMATION NO. 3519**

76743
BEYER LAW GROUP LLP / REBELVOX
P.O. BOX 1687
Cupertino, CA 95015-1687

**FILING RECEIPT**




Date Mailed: 03/11/2008

Receipt is acknowledged of this non-provisional patent application. The application will be taken up for examination in due course. Applicant will be notified as to the results of the examination. Any correspondence concerning the application must include the following identification information: the U.S. APPLICATION NUMBER, FILING DATE, NAME OF APPLICANT, and TITLE OF INVENTION. Fees transmitted by check or draft are subject to collection. Please verify the accuracy of the data presented on this receipt. **If an error is noted on this Filing Receipt, please write to the Office of Initial Patent Examination's Filing Receipt Corrections. Please provide a copy of this Filing Receipt with the changes noted thereon. If you received a "Notice to File Missing Parts" for this application, please submit any corrections to this Filing Receipt with your reply to the Notice. When the USPTO processes the reply to the Notice, the USPTO will generate another Filing Receipt incorporating the requested corrections**

**Applicant(s)**

Thomas E. Katis, Jackson, WY;
James T. Panttaja, Healdsburg, CA;
Mary G. Panttaja, Healdsburg, CA;
Matthew J. Ranney, Oakland, CA;

**Assignment For Published Patent Application**

REBELVOX, LLC, San Francisco, CA

**Power of Attorney:** The patent practitioners associated with Customer Number 76743

**Domestic Priority data as claimed by applicant**

This application is a CON of 12/028,400 02/08/2008
which claims benefit of 60/937,552 06/28/2007
and claims benefit of 60/999,619 10/19/2007

**Foreign Applications**

**If Required, Foreign Filing License Granted:** 03/07/2008

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US 12/037,256**

**Projected Publication Date:** 01/01/2009

**Non-Publication Request:** No

**Early Publication Request:** No
**** SMALL ENTITY ****

**Title**

TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS

**Preliminary Class**

379

## PROTECTING YOUR INVENTION OUTSIDE THE UNITED STATES

Since the rights granted by a U.S. patent extend only throughout the territory of the United States and have no effect in a foreign country, an inventor who wishes patent protection in another country must apply for a patent in a specific country or in regional patent offices. Applicants may wish to consider the filing of an international application under the Patent Cooperation Treaty (PCT). An international (PCT) application generally has the same effect as a regular national patent application in each PCT-member country. The PCT process **simplifies** the filing of patent applications on the same invention in member countries, but **does not result** in a grant of "an international patent" and does not eliminate the need of applicants to file additional documents and fees in countries where patent protection is desired.

Almost every country has its own patent law, and a person desiring a patent in a particular country must make an application for patent in that country in accordance with its particular laws. Since the laws of many countries differ in various respects from the patent law of the United States, applicants are advised to seek guidance from specific foreign countries to ensure that patent rights are not lost prematurely.

Applicants also are advised that in the case of inventions made in the United States, the Director of the USPTO must issue a license before applicants can apply for a patent in a foreign country. The filing of a U.S. patent application serves as a request for a foreign filing license. The application's filing receipt contains further information and guidance as to the status of applicant's license for foreign filing.

Applicants may wish to consult the USPTO booklet, "General Information Concerning Patents" (specifically, the section entitled "Treaties and Foreign Patents") for more information on timeframes and deadlines for filing foreign patent applications. The guide is available either by contacting the USPTO Contact Center at 800-786-9199, or it can be viewed on the USPTO website at http://www.uspto.gov/web/offices/pac/doc/general/index.html.

For information on preventing theft of your intellectual property (patents, trademarks and copyrights), you may wish to consult the U.S. Government website, http://www.stopfakes.gov. Part of a Department of Commerce initiative, this website includes self-help "toolkits" giving innovators guidance on how to protect intellectual property in specific countries such as China, Korea and Mexico. For questions regarding patent enforcement issues, applicants may call the U.S. Government hotline at 1-866-999-HALT (1-866-999-4158).

## LICENSE FOR FOREIGN FILING UNDER

## Title 35, United States Code, Section 184

## Title 37, Code of Federal Regulations, 5.11 & 5.15

**GRANTED**

The applicant has been granted a license under 35 U.S.C. 184, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" followed by a date appears on this form. Such licenses are issued in all applications where the conditions for issuance of a license have been met, regardless of whether or not a license may be required as

set forth in 37 CFR 5.15. The scope and limitations of this license are set forth in 37 CFR 5.15(a) unless an earlier license has been issued under 37 CFR 5.15(b). The license is subject to revocation upon written notification. The date indicated is the effective date of the license, unless an earlier license of similar scope has been granted under 37 CFR 5.13 or 5.14.

This license is to be retained by the licensee and may be used at any time on or after the effective date thereof unless it is revoked. This license is automatically transferred to any related applications(s) filed under 37 CFR 1.53(d). This license is not retroactive.

The grant of a license does not in any way lessen the responsibility of a licensee for the security of the subject matter as imposed by any Government contract or the provisions of existing laws relating to espionage and the national security or the export of technical data. Licensees should apprise themselves of current regulations especially with respect to certain countries, of other agencies, particularly the Office of Defense Trade Controls, Department of State (with respect to Arms, Munitions and Implements of War (22 CFR 121-128)); the Bureau of Industry and Security, Department of Commerce (15 CFR parts 730-774); the Office of Foreign AssetsControl, Department of Treasury (31 CFR Parts 500+) and the Department of Energy.

**NOT GRANTED**

No license under 35 U.S.C. 184 has been granted at this time, if the phrase "IF REQUIRED, FOREIGN FILING LICENSE GRANTED" DOES NOT appear on this form. Applicant may still petition for a license under 37 CFR 5.12, if a license is desired before the expiration of 6 months from the filing date of the application. If 6 months has lapsed from the filing date of this application and the licensee has not received any indication of a secrecy order under 35 U.S.C. 181, the licensee may foreign file the application pursuant to 37 CFR 5.15(b).

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: Katis et. al. | Attorney Docket No.: REBLP003/VOXP003 |
| Application No.: 12/037,256 | Examiner: Not Yet Assigned |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on April 28, 2008.

Signed: /Laura M. Dean/
Laura M. Dean

# PRELIMINARY AMENDMENT

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

Before examination of the subject application, please amend the above-identified patent application as follows:

**Amendments to the Claims** are reflected in the listing of claims, which begin on page 2 of this paper.

**Remarks/Arguments** begin on page 6 of this paper.

The listing of claims will replace all prior versions, and listings, of claims in the application:

**Listing of Claims:**

1. (Original) A communication method for communication on a first communication device, comprising:

receiving at the first communication device media of a conversation with one or more participants over a network;

storing on the first communication device the received media as the media is received over the network;

rendering on the first communication device the received media of the conversation in either a near real-time mode or a time-shifted mode; and

transitioning the rendering of the media of the conversation on the first communication device from the time-shifted mode to the near real-time mode by:

retrieving the stored media;

rendering the retrieved media from storage at an increased rendering rate;

ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the conversation as it is being received over the network; and

rendering the media of the conversation in the near real-time mode as it is being received over the network after the catch-up point.

2. (Original) The method of claim 1, wherein retrieving the stored media further comprises retrieving the stored media from a selected previous point in time in the conversation.

3. (Original) The method of claim 2, wherein rendering the retrieved media from storage further comprises rendering the stored media from the selected previous point in time in the conversation.

4. (Original) The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media at a rendering rate faster than the media was originally encoded.

5. (Original) The method of claim 4, wherein the rendering rate is variable.

6. (Original) The method of claim 5, wherein the variable rendering rate is selected at the first communication device.

7. (Original) The method of claim 1, wherein rendering the media of the conversation in the near real-time mode as it is being received over the network further comprises progressively rendering the received media as the media arrives over the network.

8. (Currently Amended) The method of claim 1, wherein rendering the received media of the conversation in the time-shifted mode further comprises:

retrieving the retrieved media from storage **or memory** after a delay; and

rendering the media retrieved from storage **or memory** after the delay.

9. (Original) The method of claim 1, wherein the rendering on the first communication device may be selectively transitioned between the near real-time mode and the time-shifted mode.

10. (Original) The method of claim 1, further comprising providing a plurality of rendering options for rendering the media, the rendering options including one or more of the following: pause, replay, play faster, play slower, jump backward, jump forward, catch up to the most recently received media, or jump to the most recently received media.

11. (Original) The method of claim 1, further comprising:

creating media on the first communication device, the created media associated with the conversation;

storing the created media; and

transmitting the created media to the one or more participants of the conversation over the network.

12. (Original) The method of claim 11, further comprising segmenting the created and the received media into messages.

13. (Original) The method of claim 12, further comprising associating the messages with the conversation.

14. (Original) The method of claim 1, further comprising mixing the media from the one or more participants of the conversation when rendering in the near real-time mode.

15. (Original) The method of claim 1, further comprising mixing the media from the one or more participants of the conversation when rendering in the time-delayed mode.

16. (Original) The method of claim 1, further comprising separating the media from the one or more participants of the conversation when rendering in the time-delayed mode so that the media from each participant of the conversation is independently rendered.

17. (Currently Amended) The method of claim 1, wherein the media comprises one or more of the following: voice, video, text, sensor data, **radio signals,** position or GPS information, or a combination thereof.

18. (Currently Amended) The method of claim 1, wherein the first communication device comprises one of the following: wherein the first communication device comprises one of the following: a ~~programmable~~ land-line phone, a ~~programmable~~ wireless **phone**, **a** cellular **phone, [[or]] a** mobile phone, a ~~programmable~~ computer, a ~~programmable~~ radio, a satellite phone, a satellite radio, a ~~programmable~~ tactical radio, or a ~~programmable~~ tactical phone.

19. (Currently Amended) The method of claim 1, wherein the network comprises a packet-based network, a circuit-based network, a cellular network, a wireless network, a radio based network, **a telephone network,** or a combination thereof.

20. (Original) The method of claim 1, wherein the media comprises one or more indexed media payloads each having a time stamp and sequence number.

21. (Original) The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media by performing one or more of the following: skipping silence in the media, skipping hesitation particles in the media, speeding up the audio content in the media, or pitch shifting.

22. (Original) The method of claim 8, wherein the delay has a duration, the duration being determined by a user of the first communication device.

## REMARKS

The Applicants have entered several amendments to the claims. No new matter has been added.

Should the Examiner believe that a telephone conference would expedite the prosecution of this application, the undersigned can be reached at the telephone number set out below.

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Reg. No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 3214592 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 28-APR-2008 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 18:34:13 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes) /Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | | REBLP003_Preliminary_Amendment.pdf | 34596<br>1817c2a104165e17af0039b0387119e6eaced706 | yes | 6 |

| | Multipart Description/PDF files in .zip description | | |
|---|---|---|---|
| | **Document Description** | **Start** | **End** |
| | Preliminary Amendment | 1 | 1 |
| | Claims | 2 | 5 |
| | Applicant Arguments/Remarks Made in an Amendment | 6 | 6 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |
| **Total Files Size (in bytes):** | 34596 |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**
**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**
**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**
**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

PTO/SB/06 (07-06)
Approved for use through 1/31/2007. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

| **PATENT APPLICATION FEE DETERMINATION RECORD**<br>Substitute for Form PTO-875 | Application or Docket Number<br>12/037,256 | Filing Date<br>02/26/2008 | ☐ To be Mailed |
|---|---|---|---|

## APPLICATION AS FILED – PART I

| | (Column 1) | (Column 2) | SMALL ENTITY ☒ | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|
| FOR | NUMBER FILED | NUMBER EXTRA | RATE ($) | FEE ($) | | RATE ($) | FEE ($) |
| ☐ BASIC FEE (37 CFR 1.16(a), (b), or (c)) | N/A | N/A | N/A | | | N/A | |
| ☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m)) | N/A | N/A | N/A | | | N/A | |
| ☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q)) | N/A | N/A | N/A | | | N/A | |
| TOTAL CLAIMS (37 CFR 1.16(i)) | minus 20 = | * | X $ = | | OR | X $ = | |
| INDEPENDENT CLAIMS (37 CFR 1.16(h)) | minus 3 = | * | X $ = | | | X $ = | |
| ☐ APPLICATION SIZE FEE (37 CFR 1.16(s)) | If the specification and drawings exceed 100 sheets of paper, the application size fee due is $250 ($125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s). | | | | | | |
| ☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j)) | | | | | | | |
| * If the difference in column 1 is less than zero, enter "0" in column 2. | | | TOTAL | | | TOTAL | |

## APPLICATION AS AMENDED – PART II

| | | (Column 1) | | (Column 2) | (Column 3) | SMALL ENTITY | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|---|---|---|
| | **04/28/2008** | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * 22 | Minus | ** 22 | = 0 | X $25 = | 0 | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * 1 | Minus | ***3 | = 0 | X $105 = | 0 | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | 0 | OR | TOTAL ADD'L FEE | |

| | | (Column 1) | | (Column 2) | (Column 3) | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| | | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * | Minus | ** | = | X $ = | | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * | Minus | *** | = | X $ = | | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | | OR | TOTAL ADD'L FEE | |

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.

** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".

*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".

The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

Legal Instrument Examiner:
/WANDA MEREDITH/

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

*If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.*

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

| APPLICATION NUMBER | FILING OR 371(C) DATE | FIRST NAMED APPLICANT | ATTY. DOCKET NO./TITLE |
|---|---|---|---|
| 12/037,256 | 02/26/2008 | Thomas E. Katis | REBLP003/VOX-P003 |

**CONFIRMATION NO. 3519**

76743
BEYER LAW GROUP LLP / REBELVOX
P.O. BOX 1687
Cupertino, CA 95015-1687

**PUBLICATION NOTICE**

*OC000000033876596*

**Title:**TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS

**Publication No.**US-2009-0003559-A1
**Publication Date:**01/01/2009

## NOTICE OF PUBLICATION OF APPLICATION

The above-identified application will be electronically published as a patent application publication pursuant to 37 CFR 1.211, et seq. The patent application publication number and publication date are set forth above.

The publication may be accessed through the USPTO's publically available Searchable Databases via the Internet at www.uspto.gov. The direct link to access the publication is currently http://www.uspto.gov/patft/.

The publication process established by the Office does not provide for mailing a copy of the publication to applicant. A copy of the publication may be obtained from the Office upon payment of the appropriate fee set forth in 37 CFR 1.19(a)(1). Orders for copies of patent application publications are handled by the USPTO's Office of Public Records. The Office of Public Records can be reached by telephone at (703) 308-9726 or (800) 972-6382, by facsimile at (703) 305-8759, by mail addressed to the United States Patent and Trademark Office, Office of Public Records, Alexandria, VA 22313-1450 or via the Internet.

In addition, information on the status of the application, including the mailing date of Office actions and the dates of receipt of correspondence filed in the Office, may also be accessed via the Internet through the Patent Electronic Business Center at www.uspto.gov using the public side of the Patent Application Information and Retrieval (PAIR) system. The direct link to access this status information is currently http://pair.uspto.gov/. Prior to publication, such status information is confidential and may only be obtained by applicant using the private side of PAIR.

Further assistance in electronically accessing the publication, or about PAIR, is available by calling the Patent Electronic Business Center at 1-866-217-9197.

Office of Data Managment, Application Assistance Unit (571) 272-4000, or (571) 272-4200, or 1-888-786-0101

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: Not Yet Assigned |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on March 9, 2009.

Signed: /Laura M. Dean/
Laura M. Dean

# INFORMATION DISCLOSURE STATEMENT
# 37 CFR §§1.56 AND 1.97(b)

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The reference(s) listed in the attached PTO Form 1449 may be material to examination of the above-identified patent application. Applicants submit a list of the identified reference(s) in compliance with their duty of disclosure pursuant to 37 CFR §§1.56 and 1.97. The Examiner is requested to make the identified reference(s) of official record in this application. The above-identified application is a continuation of prior application U.S. Patent Application No. 12/028,400. This prior application is being relied upon for an earlier filing date under 35 U.S.C. § 120. Because the listed reference(s) were either cited by the PTO, or submitted to the PTO in the prior application, under 37 CFR § 1.98(d) Applicants submit that copies need not be provided.

This Information Disclosure Statement is not to be construed as a representation that a search has been made, that additional information material to the examination of this application does not exist, or that the identified reference(s) indeed constitute prior art.

This Information Disclosure Statement is: (i) filed within three (3) months of the filing date of the above-referenced application, (ii) believed to be filed before the mailing date of a first Office Action on the merits, or (iii) believed to be filed before the mailing of a first Office

Action after the filing of a Request for Continued Examination under §1.114. Accordingly, it is believed that no fees are due in connection with the filing of this Information Disclosure Statement. However, if it is determined that any fees are due, the Commissioner is hereby authorized to charge such fees to Deposit Account 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 1 of 11 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 1. | 2001/0025377 | 8/27/01 | Hinderks | | | 12/29/00 |
| | 2. | 2002/0128029 | 9/12/02 | Nishikawa, et al. | | | 01/07/02 |
| | 3. | 2002/0150094 | 10/17/02 | Cheng, et al. | | | 10/26/01 |
| | 4. | 2002/0154745 | 10/24/02 | Shtivelman | | | 04/24/01 |
| | 5. | 2002/0159600 | 10/31/02 | Weiner | | | 07/27/01 |
| | 6. | 2002/0184368 | 12/5/02 | Wang | | | 04/06/01 |
| | 7. | 2003/0027566 | 02/06/03 | Weiner | | | 07/30/01 |
| | 8. | 2003/0028632 | 2/6/03 | Davis | | | 08/02/01 |
| | 9. | 2003/0099198 | 5/29/03 | Kiremidjian, et al. | | | 11/27/01 |
| | 10. | 2003/0126162 | 7/3/03 | Yohe, et al. | | | 01/03/02 |
| | 11. | 2003/0186722 | 10/02/03 | Weiner | | | 03/28/02 |
| | 12. | 2004/0017905 | 1/29/04 | Warrier, et al. | | | 12/17/02 |
| | 13. | 2004/0019539 | 1/29/04 | Raman, et al. | | | 12/17/02 |
| | 14. | 2004/0074448 | 4/22/04 | Bunt | | | 10/25/01 |
| | 15. | 2004/0090959 | 5/13/04 | Cinghita, et al. | | | 09/04/02 |
| | 16. | 2004/0095900 | 05/20/04 | Siegel | | | 11/14/02 |
| | 17. | 2004/0117722 | 6/17/04 | Harada | | | 11/18/03 |
| | 18. | 2004/0192353 | 9/30/04 | Mason, et al. | | | 12/23/03 |
| | 19. | 2004/0192378 | 09/30/04 | Wulkan | | | 03/25/03 |
| | 20. | 2004/0252679 | 12/16/04 | Williams et al. | | | 04/24/03 |
| | 21. | 2004/0255148 | 12/16/04 | Monteiro, et al. | | | 05/04/04 |
| | 22. | 2004/0258220 | 12/23/04 | Levine et al. | | | 03/29/02 |
| | 23. | 2005/0021819 | 01/27/05 | Kilkki | | | 01/23/04 |
| | 24. | 2005/0025308 | 02/03/05 | Tischer et al. | | | 08/30/04 |
| | 25. | 2005/0030932 | 02/10/05 | Kelly et al. | | | 09/14/04 |
| | 26. | 2005/0037706 | 2/17/05 | Settle | | | 08/01/03 |
| | 27. | 2005/0053033 | 3/10/05 | Kelly, et al. | | | 09/15/04 |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 2 of 11 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 28. | 2005/0135333 | 06/23/05 | Rojas | | | 12/18/03 |
| | 29. | 2005/0144247 | 06/30/05 | Christensen et al. | | | 12/09/03 |
| | 30. | 2005/0160345 | 7/21/05 | Walsh, et al. | | | 12/24/03 |
| | 31. | 2005/0202807 | 09/15/05 | Ayyasamy | | | 05/11/05 |
| | 32. | 2005/0207487 | 9/22/05 | Monroe | | | 02/14/05 |
| | 33. | 2005/0215228 | 09/29/05 | Fostick et al. | | | 03/26/04 |
| | 34. | 2005/0220137 | 10/6/05 | Prigent, et al. | | | 03/17/05 |
| | 35. | 2005/0259682 | 11/24/05 | Yosef, et al. | | | 02/02/01 |
| | 36. | 2005/0288101 | 12/29/05 | Lockton et al. | | | 06/24/05 |
| | 37. | 2006/0007943 | 01/12/06 | Fellman | | | 07/07/05 |
| | 38. | 2006/0045038 | 3/2/06 | Kay, et al. | | | 08/27/04 |
| | 39. | 2006/0059267 | 03/16/06 | Cugi et al. | | | 11/10/04 |
| | 40. | 2006/0059342 | 3/16/06 | Medvinsky, et al. | | | 09/16/05 |
| | 41. | 2006/0093304 | 05/04/06 | Battey et al. | | | 12/14/05 |
| | 42. | 2006/0107285 | 5/18/06 | Medvinsky | | | 11/17/05 |
| | 43. | 2006/0146822 | 07/06/06 | Kolakowski et al. | | | 12/30/04 |
| | 44. | 2006/0187897 | 8/24/06 | Dabbs et al. | | | 12/16/05 |
| | 45. | 2006/0189305 | 08/24/06 | Ando et al. | | | 02/10/06 |
| | 46. | 2006/0212582 | 9/21/06 | Gupta, et al. | | | 03/15/05 |
| | 47. | 2006/0212592 | 9/21/06 | Gupta, et al. | | | 03/15/05 |
| | 48. | 2006/0224748 | 10/5/06 | Gupta, et al. | | | 03/15/05 |
| | 49. | 2006/0244588 | 11/2/06 | Hannah, et al. | | | 03/20/06 |
| | 50. | 2006/0245367 | 11/02/06 | Jeffery et al. | | | 07/30/04 |
| | 51. | 2006/0253599 | 11/9/06 | Monteiro, et al. | | | 07/10/06 |
| | 52. | 2006/0268750 | 11/30/06 | Weiner | | | 05/02/04 |
| | 53. | 2006/0274698 | 12/7/06 | Twitchell | | | 06/05/06 |
| | 54. | 2006/0276714 | 12/7/06 | Holt, et al. | | | 06/06/05 |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 3 of 11 |

| U.S. Patent Documents | | | | | | | |
|---|---|---|---|---|---|---|---|
| **Examiner Initial** | **No.** | **Patent No.** | **Date** | **Patentee** | **Class** | **Sub-class** | **Filing Date** |
| | 55. | 2006/0282544 | 12/14/06 | Monteiro, et al. | | | 08/22/06 |
| | 56. | 2006/0288391 | 12/21/06 | Puthiyedath | | | 08/23/06 |
| | 57. | 2006/0294259 | 12/28/06 | Matefi, et al. | | | 10/24/03 |
| | 58. | 2007/0001869 | 1/4/07 | Hunzinger | | | 06/29/05 |
| | 59. | 2007/0180032 | 08/02/07 | Pearson | | | 01/27/06 |
| | 60. | 2007/0182819 | 8/9/07 | Monroe | | | 12/28/06 |
| | 61. | 2007/0207785 | 09/06/07 | Chatterjee et al. | | | 01/08/07 |
| | 62. | 2008/0000979 | 1/3/08 | Poisner | | | 06/30/06 |
| | 63. | 2008/0002621 | 1/3/08 | Ginzburg, et al. | | | 06/29/06 |
| | 64. | 2008/0002691 | 1/3/08 | Qi et al. | | | 06/29/06 |
| | 65. | 2008/0086700 | 04/10/08 | Rodriguez et al. | | | 10/06/06 |
| | 66. | 2008/0134054 | 06/05/08 | Clark et al. | | | 11/30/06 |
| | 67. | 4,807,224 | 2/21/89 | Naron, et al. | | | 08/21/87 |
| | 68. | 5,117,422 | 5/26/92 | Hauptschein, et al. | | | 07/09/90 |
| | 69. | 5,283,818 | 2/1/94 | Klausner, et al. | | | 03/31/92 |
| | 70. | 5,375,018 | 12/20/94 | Klausner, et al. | | | 07/17/91 |
| | 71. | 5,390,236 | 2/14/95 | Klausner, et al. | | | 05/12/92 |
| | 72. | 5,487,167 | 1/23/96 | Dinallo, et al. | | | 08/09/94 |
| | 73. | 5,524,140 | 6/4/96 | Klausner, et al. | | | 06/07/95 |
| | 74. | 5,572,576 | 11/5/96 | Klausner, et al. | | | 03/15/94 |
| | 75. | 5,692,213 | 11/25/97 | Goldberg et al. | | | 10/12/95 |
| | 76. | 5,734,963 | 3/31/98 | Fitzgerald, et al. | | | 06/06/95 |
| | 77. | 5,918,158 | 6/29/99 | LaPorta, et al. | | | 07/24/96 |
| | 78. | 5,963,551 | 10/5/99 | Minko | | | 11/18/96 |
| | 79. | 5,970,122 | 10/19/99 | LaPorta, et al. | | | 07/24/96 |
| | 80. | 6,037,932 | 3/14/00 | Feinleib | | | 10/7/96 |
| | 81. | 6,092,120 | 7/18/00 | Swaminathan, et al. | | | 06/26/98 |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 4 of 11 |

| U.S. Patent Documents | | | | | | | |
|---|---|---|---|---|---|---|---|
| **Examiner Initial** | **No.** | **Patent No.** | **Date** | **Patentee** | **Class** | **Sub-class** | **Filing Date** |
| | 82. | 6,104,757 | 8/15/00 | Rhee | | | 05/15/98 |
| | 83. | 6,233,389 | 5/15/01 | Barton, et al. | | | 07/30/98 |
| | 84. | 6,262,994 | 7/17/01 | Dirschedl, et al. | | | 11/14/97 |
| | 85. | 6,378,035 | 4/23/02 | Parry, et al. | | | 04/06/99 |
| | 86. | 6,480,783 | 11/12/02 | Myr | | | 03/17/00 |
| | 87. | 6,507,586 | 1/14/03 | Satran, et al. | | | 08/24/98 |
| | 88. | 6,577,599 | 6/10/03 | Gupta, et al. | | | 06/30/99 |
| | 89. | 6,580,694 | 6/17/03 | Baker | | | 08/16/99 |
| | 90. | 6,671,732 | 12/20/03 | Weiner | | | 07/24/00 |
| | 91. | 6,717,925 | 4/6/04 | Leppisaari, et al. | | | 08/10/98 |
| | 92. | 6,791,949 | 8/14/04 | Ryu, et al. | | | 04/28/00 |
| | 93. | 6,807,578 | 10/19/04 | Satran, et al. | | | 03/14/01 |
| | 94. | 6,829,473 | 12/7/04 | Raman, et al. | | | 12/17/02 |
| | 95. | 6,834,039 | 12/21/04 | Kelly, et al. | | | 11/27/00 |
| | 96. | 6,850,965 | 2/1/05 | Allen | | | 06/26/01 |
| | 97. | 6,912,544 | 06/28/05 | Weiner | | | 08/31/00 |
| | 98. | 6,970,926 | 11/29/05 | Needham, et al. | | | 10/03/00 |
| | 99. | 6,973,309 | 12/6/05 | Rygula, et al. | | | 03/14/02 |
| | 100. | 6,993,009 | 1/31/06 | Kelly, et al. | | | 12/08/00 |
| | 101. | 6,996,624 | 2/7/06 | LeCroy, et al. | | | 09/27/01 |
| | 102. | 7,002,913 | 2/21/06 | Huang, et al. | | | 01/18/01 |
| | 103. | 7,117,521 | 10/3/06 | Puthiyedath | | | 08/31/01 |
| | 104. | 7,139,371 | 11/21/06 | McElvaney | | | 12/13/05 |
| | 105. | 7,171,491 | 1/30/07 | O'Toole, et al. | | | 05/24/01 |
| | 106. | 7,187,941 | 3/6/07 | Siegel | | | 11/14/02 |
| | 107. | 7,218,709 | 7/15/04 | Garg, et al. | | | 12/22/03 |
| | 108. | 7,233,589 | 6/19/07 | Tanigawa, et al. | | | 08/30/02 |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 5 of 11 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 109. | 7,236,738 | 6/26/07 | Settle | | | 08/01/03 |
| | 110. | 7,240,105 | 7/3/07 | Satran, et al. | | | 01/26/01 |
| | 111. | 7,304,951 | 12/4/07 | Rhee | | | 11/21/01 |
| | 112. | 7,313,593 | 12/25/07 | Pulito, et al. | | | 10/24/00 |
| | 113. | 7,349,871 | 3/25/08 | Labrou, et al. | | | 07/29/03 |
| | 114. | 7,382,881 | 6/3/08 | Uusitalo, et al. | | | 06/12/03 |
| | 115. | 7,403,775 | 07/22/08 | Patel, et al. | | | 05/13/05 |

**Foreign Patent or Published Foreign Patent Application**

| Examiner Initial | | Document No. | Publication Date | Country or Patent Office | Class | Sub-Class | Translation Yes | Translation No |
|---|---|---|---|---|---|---|---|---|
| | 116. | WO 2006/114673 | 11/02/06 | PCT | | | | |
| | 117. | WO 03/073642 | 09/04/03 | PCT | | | | |

**Other Documents**

| Examiner Initial | No. | Author, Title, Place (e.g. Journal) of Publication, Date |
|---|---|---|
| | 118. | "*dircproxy*," http://home.pcisys.net/~tbc/hacks/dircproxy.htm, Downloaded on 09/26/08, 1 page. |
| | 119. | Apple Inc., "*iPhone User's Guide*," http://manuals.info.apple.com/en_US/iPhone_User_Guide.pdf, Downloaded on 10/3/08, 154 pages. |
| | 120. | Brandx.net, "*Using Talk*," http://www.brandx.net/support/usingtelnet/talk.shtml, Downloaded on 9/19/08, 2 pages. |
| | 121. | Businesswire.com "*LignUp 4.0 Delivers Industry's Most Comprehensive Suite of Communications Web Services*," http://www.businesswire.com/portal/site/google/index.jsp?ndmViewId=news_view&newsId=20070430005498&newsLang=en, Downloaded on 11/21/08, 10 pages. |
| | 122. | Calore, Michael, "*SpinVox Moves Into VoIP, Brings Voice-to-Text to Skype Users*," Monkey_Bites, http://blog.wired.com/monkeybites/2007/08/spinvox-moves-i.html, Downloaded on 10/3/08, 4 pages. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 6 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 123. | Cardei et al., "*MAC Layer QoS Support for Wireless Networks of Unmanned Air Vehicles*," Proceedings of the 37th Hawaii International Conference on System Sciences – 2004, 5-8 Jan. 2004 Page(s): 9 pp. |
| | 124. | Charny, Ben, "*Nextel pushes new 'push to talk' features*," URL: http://news.zdnet.com/2100-9584_22-134945.html, Mar 18, 2004, 3 pages. |
| | 125. | Chen et al., "*An Adaptive Multimedia Transmission Protocol for Distributed Multimedia Applications*," Proceedings of the 5th International Workshop on Multimedia Network Systems and Applications (MNSA'2003), in conjunction with The 23rd International Conference on Distributed Computing Systems (ICDCS-2003), 6 pages. |
| | 126. | Dannen, Chris, "*Technology: The Skype Mobile Phone Will Blow Your Mind*," December 18, 2007, URL: http://www.fastcompany.com/blog/chris-dannen/lab/technology-skype-mobile-phone-will-blow-your-mind, 2 pages. |
| | 127. | Erwu et al., "*Packet-late indication based (PLIB): adaptive jitter buffer*," ACM International Conference Proceeding Series; Vol. 58, Proceedings of the winter international symposium on Information and communication technologies, Cancun, Mexico, SESSION: Performance, reliability, and quality of service, Pages: 1 - 5 Year of Publication: 2004. |
| | 128. | FAQS.org, "*RFC1644 - T/TCP -- TCP Extensions for Transactions Functional S*," http://www.faqs.org/rfcs/rfc1644.html, Downloaded on 9/19/08, 26 pages. |
| | 129. | FluidVoice "Overview of FluidVoice," http://viral.media.mit.edu/wiki/tiki-index.php?page=FluidVoice, Downloaded on 9/16/08, 2 pages. |
| | 130. | GrandCentral.com, "*Call Record*," http://www.grandcentral.com/howitworks/call_record, Downloaded on 9/26/08, 1 page. |
| | 131. | GrandCentral.com, "*One Voicemail Box*," http://www.grandcentral.com/home/one_voicemail, Downloaded on 9/26/08, 1 page. |
| | 132. | GrandCentral.com, "*So Many Features, You Won't Believe it*," http://www.grandcentral.com/support/howitworks/, Downloaded on 9/26/08, 1 page. |
| | 133. | GrandCentral.com, "*Voicemail forwarding*," http://www.grandcentral.com/howitworks/voicemail_forwarding, Downloaded on 9/26/08, 1 page. |
| | 134. | Henshall, Stuart, "*HotRecorder - Record Skype Calls*," Skype Journal, URL: http://skypejournal.com/blog/archives/2005/03/hotrecorder_rec.php, March 25, 2005, 11 pages. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 7 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 135. | IRCHelp.org, "*An IRC Tutorial*," http://www.irchelp.org/irchelp/irctutorial.html, Downloaded on 09/26/08, 14 pages. |
| | 136. | Kadoink.com, "*Get a Widget*," http://www.kadoink.com/index.cfm?action=getWidgets, Downloaded on 9/19/08, 3 pages. |
| | 137. | Krishnan et al., "*EVRC-WIDEBAND: THE NEW 3GPP2 WIDEBAND VOCODER STANDARD*," IEEE International Conference on Acoustics, Speech and Signal Processing, 2007. ICASSP 2007, Publication Date: 15-20 April 2007, Volume: 2, On page(s): II-333-II-336, Honolulu, HI, |
| | 138. | Layton, Julia, "*How Slingbox Works*," 04 January 2006, HowStuffWorks.com, http://electronics.howstuffworks.com/slingbox.htm, 9 pages. |
| | 139. | LignUp.com, "*LignUp Communications Applications Server*," http://www.lignup.com/platform/index.html, Downloaded on 09/19/08, 1 page. |
| | 140. | Network Dictionary, "*Instant Message (IM) Technology Overview*," http://www.networkdictionary.com/networking/im.php, Downloaded on 9/16/08, 2 pages. |
| | 141. | Nikotalkie.com, "*Nikotalkie – Home*," http://www.nikotalkie.com/, Downloaded on 9/19/08, 2 pages. |
| | 142. | Nikotel.com, "*Click-Pop-Talk WebStart Phone*," http://www.nikotel.com/nikotel-click-pop-talk-java-phone.html, Downloaded on 9/19/08, 1 page. |
| | 143. | Notaras, George, "*dircproxy IRC Proxy*," http://www.g-loaded.eu/2007/02/01/dircproxy-irc-proxy/, Downloaded on 09/26/08, 4 pages. |
| | 144. | Pash, Adam, "*Consolidate Your Phones with GrandCentral*," http://techgroup.groups.vox.com/library/post/6a00cd978d0ef7f9cc00e398b8ff7a0002.html, Downloaded on 9/19/08, 5 pages. |
| | 145. | Patel, Nilay, "*Apple patent reveals data-to-voice translation system for cellphones*," Jul 28th 2007, URL: http://www.engadget.com/2007/07/28/apple-patent-reveals-data-to-voice-translation-system-for-cellph/, 5 pages. |
| | 146. | Piecuch et al., "*A Selective Retransmission Protocol for Multimedia on the Internet*," In Proceedings of SPIE Multimedia Systems and Applications, November 2000, Boston MA, USA, 12 pages. |
| | 147. | Qiao et al., "*SCTP Performance Issue on Path Delay Differential*," Lecture Notes in Computer Science, Springer Berlin / Heidelberg ISSN 0302-9743 (Print) 1611-3349 (Online) Volume 4517/2007, Wired/Wireless Internet Communications, Pages 43-54 Sunday, June 24, 2007. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 8 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 148. | Ramo et al., "*On comparing speech quality of various narrow- and wideband speech codecs*," Proceedings of the Eighth International Symposium on Signal Processing and Its Applications, 2005. Publication Date: August 28-31, 2005, Volume: 2, On page(s): 603- 606. |
| | 149. | Rey et al., "*I-D ACTION:draft-ietf-avt-rtp-retransmission-09.txt*," August 5, 2003, http://osdir.com/ml/ietf.avt/2003-08/msg00003.html, Downloaded on 9/19/08, 2 pages. |
| | 150. | Ribbit.com, "*Amphibian,*" http://www.ribbit.com/everyday/, Downloaded on 9/26/08, 1 page. |
| | 151. | Ribbit.com, "*Enhanced Visual Voicemail*," http://www.ribbit.com/everyday/tour/enhanced_visual_voicemail.php, Downloaded on 9/26/08, 2 pages. |
| | 152. | Ribbit.com, "*What is Ribbit? Features,*" http://www.ribbit.com/platform/features.php, Downloaded on 9/26/08, 1 page. |
| | 153. | Ribbit.com, "*What is Ribbit? Overview,*" http://www.ribbit.com/platform/index.php, Downloaded on 9/26/08, 1 page. |
| | 154. | Ribbit.com, "*What is Ribbit? Voice Architecture*," http://www.ribbit.com/platform/architecture.php, Downloaded on 9/26/08, 2 pages. |
| | 155. | Saito et al., "*IP Packet Loss Compensation Scheme with Bicast and Forwarding for Handover in Mobile Communications*," 2006 IEEE 17th International Symposium on Personal, Indoor and Mobile Radio Communications, Sept. 2006, page(s): 1-5, Helsinki. |
| | 156. | Skype.com, "*Making calls is just the start*," URL: http://www.skype.com/features/, Downloaded on 9/16/08, 2 pages. |
| | 157. | Spinvox.com, "*Home Page*," http://www.spinvox.com/, Downloaded on 9/26/08, 3 pages. |
| | 158. | Spinvox.com, "*How Does it Work?*," http://www.spinvox.com/how_it_works.html, Downloaded on 9/19/08, 2 pages. |
| | 159. | Swissvoice.net, "*PSTN*," http://www.swissvoice.net/ww/htm_ww/08_technology/content_pstn.html, Downloaded on 9/19/08, 3 pages. |
| | 160. | Tektronix, "*VOIP Technology Overview; Protocols, Convergence, Testing*," http://www.tektronics.com/voip, May 12, 2006. |
| | 161. | The Jabber Extensible Communications Platform™, "*Products // Jabber XCP*," URL: http://www.jabber.com/CE/JabberXCP, Downloaded on 9/16/08, 2 pages. |

| **Examiner** | **Date Considered** |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 9 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 162. | ThunkDifferent.com, "*YouMail vs. Google Grandcentral Voice Mail Service*," http://thunkdifferent.com/2007/10/11/youmail-vs-google-grandcentral-voice-mail-service/, Downloaded on 10/03/08, 6 pages. |
| | 163. | VOIP-News.com, "*Company Profile, LignUp,* " http://www.voip-news.com/vendors/lignup/, Downloaded on 12/5/08, 6 pages. |
| | 164. | WikiBooks, "*Internet Technologies/IRC*," http://en.wikibooks.org/wiki/Internet_Technologies/IRC, Downloaded on 9/19/08, 4 pages. |
| | 165. | WikiPedia – The Free Encyclopedia, "*E-mail*," http://en.wikipedia.org/wiki/E-mail, Downloaded on 9/19/08, 8 pages. |
| | 166. | WikiPedia – The Free Encyclopedia, "*Internet Relay Chat*," http://en.wikipedia.org/wiki/Internet_Relay_Chat, Downloaded on 10/03/08, 11 pages. |
| | 167. | WikiPedia – The Free Encyclopedia, "*Spinvox*," http://en.wikipedia.org/wiki/Spinvox, Downloaded on 9/26/08, 1 page. |
| | 168. | WikiPedia – The Free Encyclopedia, "*TiVo*", http://en.wikipedia.org/wiki/TiVo, Downloaded on 09/16/08, 6 pages. |
| | 169. | Yavuz et al., "*VoIP over cdma2000 1xEV-DO Revision* A," IEEE Communications Magazine, February 2006, pp 88-95. |
| | 170. | HotRecorder.com, "*Features*," http://www.hotrecorder.com/music_features.asp, downloaded on 9/26/08, 1 page. |
| | 171. | HotRecorder.com, "*Help*," http://www.hotrecorder.com/music_help.asp, downloaded on 9/26/08, 3 pages. |
| | 172. | HotRecorder.com, "*FAQs*," http://www.hotrecorder.com/music_support.asp, downloaded on 9/26/08, 1 page. |
| | 173. | WikiPedia – The Free Encyclopedia, "*Skype,* " http://en.wikipedia.org/wiki/Skype, Downloaded on 09/26/08, 7 pages. |
| | 174. | WikiPedia – The Free Encyclopedia, "*Skype Protocol,* " http://en.wikipedia.org/wiki/Skype, Downloaded on 09/26/08, 4 pages. |
| | 175. | Jabber.org, "*Main page*," http://www.jabber.org/web/main_page, Sept. 2, 2008, downloaded on 9/26/08, 1 page. |
| | 176. | Jabber.org, "*FAQ*," http://www.jabber.org/web/faq, Sept. 4, 2008, downloaded on 9/26/08, 2 pages. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 10 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 177. | Apple Inc., "*iPhone: About Visual Voicemail*," http://www.support.apple.com/kb/HT1486, Downloaded on 09/26/08, 3 pages. |
| | 178. | Jabber.org, "*Products // Jabber XCP // Benefits*," http://www.jabber.com/CE/JabberXCPBenefits, downloaded on 9/26/08, 1 page. |
| | 179. | Jabber.org, "*Products // Jabber Clients*," http://www.jabber.com/CE/JabberClients, downloaded on 9/26/08, 1 page. |
| | 180. | Jabber.org, "*Products // JabberNow*" http://www.jabber.com/CE/JabberNow, downloaded on 9/26/08, 1 page. |
| | 181. | KillerStartups.com, "*Kadoink.com - Mobilizing your Socializing*," http://www.killerstartups.com/Web20/kadoink-com-moblizing-your-socializing, Downloaded on 09/26/08, 3 pages. |
| | 182. | CNETNews.com, "*Kadoink's phonecast lines officially open for texting*," http://news.cnet.com/8301-17939_109-9829877-2.html, downloaded on 9/26/08, 1 page. |
| | 183. | BusinessWire.com, "*LignUp 4.0 Delivers Industry's Most Comprehensive Suite of Communications Web Services*," April 30, 2007, http://www.businesswire.com/portal/site/google/?ndmViewId=news_view&newsId=20070430005498&newsLang=en, Downloaded on 09/26/08, 6 pages. |
| | 184. | VOIP-News.com, "*Company Profile – LignUp*," http://www.voip-news.com/vendors/lignup/, Downloaded on 10/03/08, 6 pages. |
| | 185. | JustAnotheriPhoneBlog.com, "*Nikotalkie - Just Talk, Don't Type*," http://justanotheriphoneblog.com/wordpress/2007/10/13/nikotalkie-just-talk-dont-type/, Downloaded on 09/26/08, 10 pages. |
| | 186. | WikiPedia – The Free Encyclopedia, "*Push To Talk*" http://en.wikipedia.org/wiki/Push_to_talk, Downloaded on 09/26/08, 3 pages. |
| | 187. | WikiPedia – The Free Encyclopedia, "*Slingbox*" http://en.wikipedia.org/wiki/Slingbox, Downloaded on 09/26/08, 4 pages. |
| | 188. | About.com, "*Linux / Unix Command: talk*,"http://linux.about.com/od/commands/l/blcmdl1_talk.htm, Downloaded on 09/26/08, 2 pages. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure | Applicant | KATIS et al. |
| Statement By Applicant | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 11 of 11 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 189. | Fall, Kevin, "*A Delay-Tolerant Network Architecture for Challenged Internets*," February 2003, http://www.dtnrg.org/docs/papers/IRB-TR-03-003.pdf, 15 pages. |
| | 190. | Chuah et al., "*Store-and-Forward Performance in a DTN*," Vehicular Technology Conference, 2006. VTC 2006-Spring. IEEE 63rd, Publication Date: 7-10 May 2006, Volume: 1, On page(s): 187-191. |
| | 191. | Krishnan, et al., "*The SPINDLE Disruption-Tolerant Networking System*," Military Communications Conference, 2007. MILCOM 2007. IEEE Volume , Issue , 29-31 Oct. 2007 Page(s):1 – 7. |
| | 192. | WikiPedia – The Free Encyclopedia, "*Visual Voicemail*" http://en.wikipedia.org/wiki/Visual_voicemail, downloaded on 9/26/08, 1 page. |
| | 193. | Amir et al., "*An Overlay Architecture for High Quality VoIP Streams*,", IEEE Transactions on Multimedia, Publication Date: Dec. 2006, Volume: 8, Issue:6, On page(s): 1250-1262. |
| | 194. | Rothermel et al., "*An Adaptive Stream Synchronization Protocol*," Lecture Notes In Computer Science; Vol. 1018, Proceedings of the 5th International Workshop on Network and Operating System Support for Digital Audio and Video, Pages: 178 – 189, Year of Publication: 1995. |
| | 195. | Baset et al., "*An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol*," INFOCOM 2006. 25th IEEE International Conference on Computer Communications. Proceedings (2006), pp. 1-11. |
| | 196. | Cerf et al., "*A Protocol for Packet Network Intercommunication*," Communications, IEEE Transactions on, Volume 22, Issue 5, May 1974 Page(s): 637 – 648. |
| | 197. | International Search Report from PCT/US2008/062941, mailed October 24, 2008. |
| | 198. | Written Opinion from PCT/US2008/062941, mailed October 24, 2008. |
| | 199. | International Search Report from PCT/US2008/062049, mailed January 12, 2009. |
| | 200. | Written Opnion from PCT/US2008/062049, mailed January 12, 2009. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 4924290 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 09-MAR-2009 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 16:45:51 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Information Disclosure Statement Letter | IDS_Regular_Transmittal.pdf | 24338<br>30e5d37e43ea58b9843a6628e2ddab6d18b4cf7a | no | 2 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |

| | | | | | |
|---|---|---|---|---|---|
| 2 | Information Disclosure Statement (IDS) Filed (SB/08) | 1449_for_REBL_Global_IDS.pdf | 116852<br>8a9f00397f2a3bfbb9b81b7a7af6740fb88131f9 | no | 11 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |
| This is not an USPTO supplied IDS fillable form | |
| **Total Files Size (in bytes):** | 141190 |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**

**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**

**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**

**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

| Form 1449 (Modified) | Atty Docket No. | REBLP003/VOX-P003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | 02/26/08 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 1 of 3 |

| U.S. Patent Documents | | | | | | | |
|---|---|---|---|---|---|---|---|
| **Examiner Initial** | **No.** | **Patent No.** | **Date** | **Patentee** | **Class** | **Sub-class** | **Filing Date** |
| | 1. | 2009/0175425 | 07/09/09 | Lee | | | 01/03/08 |
| | 2. | 2005/0102358 | 05/12/05 | Gold et al. | | | 11/09/04 |
| | 3. | 2009/0063698 | 03/05/09 | Xu et al. | | | 09/04/07 |
| | 4. | 6,807,565 | 10/19/04 | Dodrill et al. | | | 06/27/00 |
| | 5. | 6,931,114 | 08/16/05 | Martin | | | 12/22/00 |
| | 6. | 7,039,040 | 05/02/06 | Burg | | | 06/07/99 |
| | 7. | 7,039,675 | 05/02/06 | Kato | | | 06/30/99 |
| | 8. | 7,047,030 | 05/16/06 | Forsyth | | | 05/02/02 |
| | 9. | 7,111,044 | 09/19/06 | Lee | | | 11/12/02 |
| | 10. | 7,305,438 | 12/04/07 | Christensen et al. | | | 12/09/03 |
| | 11. | 7,626,951 | 12/01/09 | Croy et al. | | | 08/15/06 |
| | 12. | 5,737,011 | 04/07/98 | Lukacs | | | 05/03/95 |
| | 13. | 6,175,619 | 01/16/01 | DeSimone | | | 01/16/01 |
| | 14. | 6,564,261 | 05/13/03 | Gudjonsson et al. | | | 05/09/00 |
| | 15. | 6,721,703 | 04/13/04 | Jackson et al. | | | 03/02/01 |
| | 16. | 2009/0003544 | 01/01/09 | Katis et al. | | | 03/05/08 |
| | 17. | 2009/0003545 | 01/01/09 | Katis et al. | | | 04/14/08 |
| | 18. | 2009/0003563 | 01/01/09 | Katis et al. | | | 04/15/08 |
| | 19. | 2009/0003557 | 01/01/09 | Katis et al. | | | 04/25/08 |

| Foreign Patent or Published Foreign Patent Application | | | | | | | | |
|---|---|---|---|---|---|---|---|---|
| **Examiner Initial** | **No.** | **Document No.** | **Publication Date** | **Country or Patent Office** | **Class** | **Sub-Class** | **Translation Yes** | **No** |
| | 20. | WO 2007/026320 | 03/08/07 | PCT | | | | |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003/VOX-P003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | 02/26/08 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 2 of 3 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 21. | WikiPedia – The Free Encyclopedia, "*Eudora (email client)*," http://en.wikipedia.org/wiki/Eudora_(e-mail_client), Downloaded on 08/20/09, 3 pages. |
| | 22. | "*Eudora*," Answers.com, http://www.answers.com/topic/eudora-e-mail-client, Downloaded on 08/20/09, 4 pages. |
| | 23. | "*The Eudora Open Messaging Advantage*," Qualcomm, 1997, Part Number 100-50030-1, 23 pages. |
| | 24. | "*Aspera – Next Generation File Transport – Broadcasting & Entertainment Media*," Asperasoft.com, http://www.asperasoft.com/en/industries/digital_media_10/Broadcast_Entertainment_Media_5, Downloaded on 09/22/09, 2 pages. |
| | 25. | "*Aspera – Next Generation File Transport – fasp™ transfer times*," Asperasoft.com, http://www.asperasoft.com/en/technology/fasp_transfer_times_14/fasp_transfer_times_14, Downloaded on 09/22/09, 1 page. |
| | 26. | "*Aspera – Next Generation File Transport – the fasp solution*," Asperasoft.com, http://www.asperasoft.com/en/technology/fasp_solution_3/the_fasp_solution_3, Downloaded on 09/22/09, 3 pages. |
| | 27. | "*Aspera – Next Generation File Transport – the shortcomings of TCP file transfer*," Asperasoft.com, http://www.asperasoft.com/en/technology/shortcomings_of_TCP_2/the_shortcomings_of_TCP_file_transfer_2, Downloaded on 09/22/09, 2 pages. |
| | 28. | "*Aspera – Next Generation File Transport – fasp technology overview*" Asperasoft.com, http://www.asperasoft.com/en/technology/fasp_overview_1/fasp_technology_overview_1, Downloaded on 09/22/09, 2 pages. |
| | 29. | "*Aspera fasp™ High Speed Transport- A Critical Technology Comparison,*" White Paper, Asperasoft.com, http://www.asperasoft.com/en/technology/white_papers_13/aspera_fasp_high_speed_transport_13, Downloaded on 09/22/09, 11 pages. |
| | 30. | "*Palringo Brings First Push-to-Talk Application to the iPhone*," RedOrbit.com, http://www.redorbit.com/news/technology/1525545/palringo_brings_first_pushtotalk_application_to_the_iphone/index.html, Downloaded on 08/13/2009, 2 pages. |
| | 31. | *Palringo – Features*," Palringo.com, http://www.palringo.com/en/gb/features/, Downloaded on 08/13/2009, 1 page. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003/VOX-P003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | 02/26/08 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 3 of 3 |

| Other Documents | | |
|---|---|---|
| **Examiner Initial** | **No.** | **Author, Title, Place (e.g. Journal) of Publication, Date** |
| | 32. | Moren, Dan, "*Palringo Brings Picture Messaging to Iphone*," http://www.pcworld.com/article/149108/palringo_brings_picture_messaging_to_iphone.html, Downloaded on 08/13/2009, 3 pages. |
| | 33. | Paul, Ryan, "*Gmail gets Google Talk integration*,"Arstechnica.com, http://arstechnica.com/old/content/2006/02/6128.ars , Downloaded on 08/20/09, 1 page. |
| | 34. | Sherman, Chris, "*Google Integrates Chat with Gmail*,"Search Engine Watch, http://searchenginewatch.com/3583121, February 7, 2006, Downloaded on 08/20/2009, 2 pages. |
| | 35. | "*About Gmail*," http://mail.google.com/mail/help/chat.html, Downloaded on 08/20/09, 3 pages. |
| | 36. | WikiPedia – The Free Encyclopedia, "*Google Talk*," http://en.wikipedia.org/wiki/Google_Talk, Downloaded on 08/20/09, 8 pages. |
| | 37. | Azuri, Calvin, "*Palringo Gold Launched on BlackBerry Smartphone*", April 20, 2009, http://ivr.tmcnet.com/topics/ivr-voicexml/articles/54573-palringo-gold-launched-blackberry-smartphone.htm, Downloaded on 08/13/2009, 3 pages. |
| | 38. | Liaw, Kim Poh, "*Palringo launches its IM Software for Android Phones*," Slashphone.com, March 24, 2009, http://www.slashphone.com/palringo-launches-its-im-software-for-android-phones-245111, Downloaded on 08/13/2009, 8 pages. |
| | 39. | WikiPedia – The Free Encyclopedia, "*Palringo*" http://en.wikipedia.org/wiki/Palringo, Downloaded on 08/13/09, 1 page. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 6708090 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Kathryn Throckmorton |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 24-DEC-2009 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 14:44:02 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Transmittal Letter | Global2IDSRegularTransmittal.pdf | 24341<br>7b0a475f94dbd90e7a6c0dbe5123c814a5b5c32d | no | 2 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |

| | | | | | |
|---|---|---|---|---|---|
| 2 | Information Disclosure Statement (IDS) Filed (SB/08) | 1449forREBLGlobalIDS_2.pdf | 46982<br>70619eb34aeb4fe04d2823bd1425a6c5b3bd1cf5 | no | 3 |

**Warnings:**

**Information:**

This is not an USPTO supplied IDS fillable form

| | |
|---|---|
| **Total Files Size (in bytes):** | 71323 |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**

**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**

**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**

**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: Unassigned |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION

I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on December 24, 2009.

Signed: /Kathryn Throckmorton/
Kathryn Throckmorton

# INFORMATION DISCLOSURE STATEMENT
# 37 CFR §§1.56 AND 1.97(b)

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The reference(s) listed in the attached PTO Form 1449 may be material to examination of the above-identified patent application. Applicants submit a list of the identified reference(s) in compliance with their duty of disclosure pursuant to 37 CFR §§1.56 and 1.97. The Examiner is requested to make the identified reference(s) of official record in this application. The above-identified application is a continuation of prior application U.S. Patent Application No. 12/028,400. This prior application is being relied upon for an earlier filing date under 35 U.S.C. § 120. Because the listed reference(s) were either cited by the PTO, or submitted to the PTO in the prior application, under 37 CFR § 1.98(d) Applicants submit that copies need not be provided.

This Information Disclosure Statement is not to be construed as a representation that a search has been made, that additional information material to the examination of this application does not exist, or that the identified reference(s) indeed constitute prior art.

This Information Disclosure Statement is: (i) filed within three (3) months of the filing date of the above-referenced application, (ii) believed to be filed before the mailing date of a first Office Action on the merits, or (iii) believed to be filed before the mailing of a first Office

Action after the filing of a Request for Continued Examination under §1.114. Accordingly, it is believed that no fees are due in connection with the filing of this Information Disclosure Statement. However, if it is determined that any fees are due, the Commissioner is hereby authorized to charge such fees to Deposit Account 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et. al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: KUNTZ, CURTIS A |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on May 6, 2010.

Signed: /Laura M. Dean/
Laura M. Dean

# SECOND PRELIMINARY AMENDMENT

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

Before examination of the subject application, please amend the above-identified patent application as follows:

**Amendments to the Claims** are reflected in the listing of claims, which begin on page 2 of this paper.

**Remarks** begin on page 9 of this paper.

The listing of claims will replace all prior versions, and listings, of claims in the application:

**Listing of Claims:**

1. (Currently Amended) A communication method for communication on a first communication device, comprising:

receiving at the first communication device media of a conversation ~~with one or more participants~~ over a network;

storing on the first communication device the received media as the media is received over the network; **and**

~~rendering on the first communication device the received media of the conversation in either a near real-time mode or a time-shifted mode; and~~

transitioning **[[the]]** rendering of the media of the conversation on the first communication device from **[[the]]** **a** time-shifted mode to ~~the near~~ **a** real-time mode by:

retrieving the stored media;

rendering the retrieved media from storage at an increased rendering rate;

ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the conversation as it is being received over the network; and

rendering the media of the conversation in the **[[near]]** real-time mode as it is being received over the network after the catch-up point.

2. (Original) The method of claim 1, wherein retrieving the stored media further comprises retrieving the stored media from a selected previous point in time in the conversation.

3. (Original) The method of claim 2, wherein rendering the retrieved media from storage further comprises rendering the stored media from the selected previous point in time in the conversation.

4. (Original) The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media at a rendering rate faster than the media was originally encoded.

5. (Original) The method of claim 4, wherein the rendering rate is variable.

6. (Original) The method of claim 5, wherein the variable rendering rate is selected at the first communication device.

7. (Currently Amended) The method of claim 1, wherein rendering the media of the conversation in the **[[near]]** real-time mode as it is being received over the network further comprises progressively rendering the received media as the media **progressively** arrives over the network.

8. (Currently Amended) The method of claim 1, wherein rendering the received media of the conversation in the time-shifted mode further comprises:

retrieving the retrieved media from storage or memory **an arbitrary time** after ~~a delay~~ **the media was received and stored**; and

rendering the media retrieved from storage or memory ~~after the delay~~.

9. (Currently Amended) The method of claim 1, wherein the rendering **of the conversation** on the first communication device may be selectively transitioned between the **[[near]]** real-time mode and the time-shifted mode.

10. (Original) The method of claim 1, further comprising providing a plurality of rendering options for rendering the media, the rendering options including one or more of the following: pause, replay, play faster, play slower, jump backward, jump forward, catch up to the most recently received media, or jump to the most recently received media.

11. (Currently Amended) The method of claim 1, further comprising:

creating media on the first communication device, the created media associated with the conversation;

storing the created media; and

transmitting the created media to **[[the]]** one or more participants of the conversation over the network.

12. (Currently Amended) The method of claim **[[11]]** **1**, further comprising segmenting the ~~created and the received~~ media **of the conversation** into messages.

13. (Original) The method of claim 12, further comprising associating the messages with the conversation.

14. (Currently Amended) The method of claim 1, further comprising mixing the media from **[[the]]** one or more participants of the conversation when rendering ~~**in the near real-time mode**~~ **the media of the conversation**.

15. (Cancelled)

16. (Currently Amended) The method of claim 1, further comprising separating the media from **[[the]]** one or more participants of the conversation ~~**when rendering in the time-delayed mode**~~ so that the media from **[[each]]** **a selected** participant of the conversation is independently rendered.

17. (Previously Presented) The method of claim 1, wherein the media comprises one or more of the following: voice, video, text, sensor data, radio signals, position or GPS information, or a combination thereof.

18. (Previously Presented) The method of claim 1, wherein the first communication device comprises one of the following: a land-line phone, a wireless phone, cellular phone, a mobile phone, a computer, a radio, a satellite phone, a satellite radio, a tactical radio, or a tactical phone.

19. (Previously Presented) The method of claim 1, wherein the network comprises a packet-based network, a circuit-based network, a cellular network, a wireless network, a radio based network, a telephone network, or a combination thereof.

20. (Currently Amended) The method of claim 1, wherein the media comprises one or more indexed media payloads ~~**each having a time stamp and sequence number**~~.

21. (Original) The method of claim 1, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media by performing one or more of the following: skipping silence in the media, skipping hesitation particles in the media, speeding up the audio content in the media, or pitch shifting.

22. (Currently Amended) The method of claim 8, wherein the ~~delay has a duration, the duration being determined~~ **arbitrary time is defined** by a user of the first communication device.

23. (New) A method, comprising:

receiving over a network media of a message at a communication device;

progressively storing the media of the message as the media of the message is received over the network; and

rendering the media of the message out of storage, starting at a previous point in time of the message relative to the current point of the message, as the media of the message is being received over the network at the communication device.

24. (New) The method of claim 23, wherein the rendering of the media of the message from storage further comprises rendering the media of the message from storage at a faster rate than the media of the message was originally encoded.

25. (New) The method of claim 24, further comprising:

ascertaining a catch up point where the rendering of the media out of storage at the faster rate catches up to and coincides with the media of the message as the media of the message is received over the network; and

seamlessly transitioning the rendering the media of the message from out of storage to the media received over the network after ascertaining the catch up point.

26. (New) The method of claim 25, wherein rendering after the catch up point further comprises the seamless transition of the media of the message from a time-shifted mode to a real-time mode.

27. (New) The method of claim 23, further comprising providing a plurality of rendering options for rendering the media of the message on the communication device, the rendering options including one or more of the following: pause, replay, play faster, play slower, jump backward, jump forward, catch up to the most recently received media, or jump to the most recently received media.

28. (New) The method of claim 23, wherein the media of the message comprises one or more indexed media payloads.

29. (New) The method of claim 23, further comprising associating the message with a conversation.

30. (New) The method of claim 29, further comprising:

creating one or more messages using the communication device, the created messages associated with the conversation; and

transmitting over the network the created one or more messages from the communication device to one or more other participants of the conversation.

31. (New) The method of claim 23, wherein rendering the media out of storage further comprises rendering the media out of storage by performing one or more of the following: skipping silence gaps in the media, skipping hesitation particles in the media, speeding up the audio content in the media, or pitch shifting.

32. (New) The method of claim 23, wherein the media of the message comprises one or more of the following: voice, video, text, sensor data, radio signals, position or GPS information, or a combination thereof.

33. (New) The method of claim 23, wherein the communication device comprises one of the following: a land-line phone, a wireless phone, cellular phone, a mobile phone, a computer, a radio, a satellite phone, a satellite radio, a tactical radio, or a tactical phone.

34. (New) The method of claim 23, wherein the network comprises a packet-based network, a circuit-based network, a cellular network, a wireless network, a radio based network, a telephone network, or a combination thereof.

35. (New) A communication method for communication on a communication device, comprising:

receiving at the communication device media of a message over a network;

storing on the communication device the received media of the message as the media is received over the network; and

transitioning the rendering of the media of the message on the communication device from a time-shifted mode to a real-time mode by:

retrieving the stored media of the message;

rendering the retrieved media from storage at an increased rendering rate;

ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the message as it is being received over the network; and

rendering the media of the message in the real-time mode as it is being received over the network after the catch-up point.

36. (New) The method of claim 35, wherein rendering the retrieved media from storage further comprises rendering the stored media starting from a selected previous point in time in the message, the previous point in time relative to the current point of the message as the media is received at the communication device over the network.

37. (New) The method of claim 36, wherein rendering the retrieved media from storage at the increased rendering rate further comprises rendering the retrieved media at a rendering rate faster than the media was originally encoded.

38. (New) The method of claim 35, wherein rendering the received media of the message in the time-shifted mode further comprises:

retrieving the retrieved media from storage or memory an arbitrary time after the media was stored; and

rendering the media retrieved from storage or memory.

39. (New) The method of claim 1, wherein the message is associated with a conversation and the rendering of the conversation on the first communication device is seamlessly transitioned from the time-shifted mode to the real-time mode after the catch up point.

40. (New) The method of claim 35, wherein rendering the media from storage further comprises rendering the media from storage by performing one or more of the following: skipping silence gaps in the media, skipping hesitation particles in the media, speeding up the audio content in the media, or pitch shifting.

41. (New) The method of claim 35, wherein the media of the message comprises one or more of the following: voice, video, text, sensor data, radio signals, position or GPS information, or a combination thereof.

42. (New) The method of claim 35, wherein the communication device comprises one of the following: a land-line phone, a wireless phone, cellular phone, a mobile phone, a computer, a radio, a satellite phone, a satellite radio, a tactical radio, or a tactical phone.

43. (New) The method of claim 35, wherein the network comprises a packet-based network, a circuit-based network, a cellular network, a wireless network, a radio based network, a telephone network, or a combination thereof.

## REMARKS

The Applicants have entered several amendments to the claims. No new matter has been added.

Should the Examiner believe that a telephone conference would expedite the prosecution of this application, the undersigned can be reached at the telephone number set out below.

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Reg. No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

In re application of: KATIS et. al.

Application No.: 12/037,256

Filed: February 26, 2008

Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS

Attorney Docket No.: REBLP003/VOX-P003

Examiner: KUNTZ, CURTIS A

Group: 2614

Confirmation No.: 3519

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on May 6, 2010.

Signed: /Laura M. Dean/
Laura M. Dean

# AMENDMENT TRANSMITTAL

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Sir:

Transmitted herewith is an Amendment in the above-identified application.
The fee has been calculated as shown below.

| | Claims After Amendment | | Highest Previously Paid For | Present Extra | Small Entity Rate Fee | Large Entity Rate Fee |
|---|---|---|---|---|---|---|
| Total Claims | 42 | MINUS | 22 | 20 | x 26 = 520.00 | x 52 = 00 |
| Independent Claims | 3 | MINUS | 03 | 00 | x 110 = 00 | x 220 = 00 |
| Multiple Dependent Claim Present and Fee Not Previously Paid | | | | | | |
| | | | | Total | $520.00 | $0.00 |

☒ Applicant(s) believe that no (additional) Extension of Time is required; however, if it is determined that such an extension is required, Applicant(s) hereby petition that such an extension be granted and authorize the Commissioner to charge the required fees for an Extension of Time under 37 CFR 1.136 to Deposit Account No. 50-4481.

☒ Please charge the required fees, or any additional fees required to facilitate filing the enclosed response, to Deposit Account No. 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Reg. No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687

# Electronic Patent Application Fee Transmittal

| | |
|---|---|
| **Application Number:** | 12037256 |
| **Filing Date:** | 26-Feb-2008 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |

Filed as Small Entity

## Utility under 35 USC 111(a) Filing Fees

| Description | Fee Code | Quantity | Amount | Sub-Total in USD($) |
|---|---|---|---|---|
| **Basic Filing:** | | | | |
| **Pages:** | | | | |
| **Claims:** | | | | |
| Claims in excess of 20 | 2202 | 20 | 26 | 520 |
| **Miscellaneous-Filing:** | | | | |
| **Petition:** | | | | |
| **Patent-Appeals-and-Interference:** | | | | |
| **Post-Allowance-and-Post-Issuance:** | | | | |
| **Extension-of-Time:** | | | | |

| Description | Fee Code | Quantity | Amount | Sub-Total in USD($) |
|---|---|---|---|---|
| **Miscellaneous:** | | | | |
| | | | **Total in USD ($)** | **520** |

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 7563400 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 06-MAY-2010 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 18:56:58 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | yes |
| Payment Type | Deposit Account |
| Payment was successfully received in RAM | $520 |
| RAM confirmation Number | 4511 |
| Deposit Account | 504481 |
| Authorized User | |

The Director of the USPTO is hereby authorized to charge indicated fees and credit any overpayment as follows:

Charge any Additional Fees required under 37 C.F.R. Section 1.16 (National application filing, search, and examination fees)

Charge any Additional Fees required under 37 C.F.R. Section 1.17 (Patent application and reexamination processing fees)

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | | Prelim_Amend_2.pdf | 45784<br>7da363d206d2654a60eb3ea6cf58ed7090157540 | yes | 9 |

| Multipart Description/PDF files in .zip description | | |
|---|---|---|
| **Document Description** | **Start** | **End** |
| Preliminary Amendment | 1 | 1 |
| Claims | 2 | 8 |
| Applicant Arguments/Remarks Made in an Amendment | 9 | 9 |

**Warnings:**

**Information:**

| | | | | | |
|---|---|---|---|---|---|
| 2 | Miscellaneous Incoming Letter | Prelim_Amendment_Transmittal_2.pdf | 22247<br>74802123a5349f373ea6de55e7ba36fc22acd7c6 | no | 1 |

**Warnings:**

**Information:**

| | | | | | |
|---|---|---|---|---|---|
| 3 | Fee Worksheet (PTO-875) | fee-info.pdf | 30333<br>7c635df7a3d3de1c5107a6972851ca24f59ac3e1 | no | 2 |

**Warnings:**

**Information:**

| Total Files Size (in bytes): | 98364 |
|---|---|

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**

**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**

**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**

**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

PTO/SB/06 (07-06)
Approved for use through 1/31/2007. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

| **PATENT APPLICATION FEE DETERMINATION RECORD**<br>Substitute for Form PTO-875 | Application or Docket Number<br>12/037,256 | Filing Date<br>02/26/2008 | ☐ To be Mailed |
|---|---|---|---|

## APPLICATION AS FILED – PART I

| | (Column 1) | (Column 2) | SMALL ENTITY ☒ | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|
| FOR | NUMBER FILED | NUMBER EXTRA | RATE ($) | FEE ($) | | RATE ($) | FEE ($) |
| ☐ BASIC FEE (37 CFR 1.16(a), (b), or (c)) | N/A | N/A | N/A | | | N/A | |
| ☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m)) | N/A | N/A | N/A | | | N/A | |
| ☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q)) | N/A | N/A | N/A | | | N/A | |
| TOTAL CLAIMS (37 CFR 1.16(i)) | minus 20 = | * | X $ = | | OR | X $ = | |
| INDEPENDENT CLAIMS (37 CFR 1.16(h)) | minus 3 = | * | X $ = | | | X $ = | |
| ☐ APPLICATION SIZE FEE (37 CFR 1.16(s)) | If the specification and drawings exceed 100 sheets of paper, the application size fee due is $250 ($125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s). | | | | | | |
| ☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j)) | | | | | | | |
| * If the difference in column 1 is less than zero, enter "0" in column 2. | | | TOTAL | | | TOTAL | |

## APPLICATION AS AMENDED – PART II

| | | (Column 1) | | (Column 2) | (Column 3) | SMALL ENTITY | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|---|---|---|
| | **05/06/2010** | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * 42 | Minus | ** 22 | = 20 | X $26 = | 520 | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * 3 | Minus | ***3 | = 0 | X $110 = | 0 | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | 520 | OR | TOTAL ADD'L FEE | 0 |

| | | (Column 1) | | (Column 2) | (Column 3) | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| | | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * | Minus | ** | = | X $ = | | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * | Minus | *** | = | X $ = | | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | | OR | TOTAL ADD'L FEE | |

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.

** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".

*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".

The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

Legal Instrument Examiner:
/EAON SMALLWOOD/

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

*If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.*

PTO/SB/06 (07-06)
Approved for use through 1/31/2007. OMB 0651-0032
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE
Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

| **PATENT APPLICATION FEE DETERMINATION RECORD**<br>Substitute for Form PTO-875 | Application or Docket Number<br>12/037,256 | Filing Date<br>02/26/2008 | ☐ To be Mailed |
|---|---|---|---|

## APPLICATION AS FILED – PART I

| | (Column 1) | (Column 2) | SMALL ENTITY ☒ | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|
| FOR | NUMBER FILED | NUMBER EXTRA | RATE ($) | FEE ($) | | RATE ($) | FEE ($) |
| ☐ BASIC FEE (37 CFR 1.16(a), (b), or (c)) | N/A | N/A | N/A | | | N/A | |
| ☐ SEARCH FEE (37 CFR 1.16(k), (i), or (m)) | N/A | N/A | N/A | | | N/A | |
| ☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q)) | N/A | N/A | N/A | | | N/A | |
| TOTAL CLAIMS (37 CFR 1.16(i)) | minus 20 = | * | X $ = | | OR | X $ = | |
| INDEPENDENT CLAIMS (37 CFR 1.16(h)) | minus 3 = | * | X $ = | | | X $ = | |
| ☐ APPLICATION SIZE FEE (37 CFR 1.16(s)) | If the specification and drawings exceed 100 sheets of paper, the application size fee due is $250 ($125 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s). | | | | | | |
| ☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j)) | | | | | | | |
| * If the difference in column 1 is less than zero, enter "0" in column 2. | | | TOTAL | | | TOTAL | |

## APPLICATION AS AMENDED – PART II

| | | (Column 1) | | (Column 2) | (Column 3) | SMALL ENTITY | | OR | OTHER THAN SMALL ENTITY | |
|---|---|---|---|---|---|---|---|---|---|---|
| | **05/06/2010** | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * 42 | Minus | ** 22 | = 20 | X $26 = | 520 | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * 3 | Minus | ***3 | = 0 | X $110 = | 0 | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | **520** | OR | TOTAL ADD'L FEE | **0** |

| | | (Column 1) | | (Column 2) | (Column 3) | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| | | CLAIMS REMAINING AFTER AMENDMENT | | HIGHEST NUMBER PREVIOUSLY PAID FOR | PRESENT EXTRA | RATE ($) | ADDITIONAL FEE ($) | | RATE ($) | ADDITIONAL FEE ($) |
| AMENDMENT | Total (37 CFR 1.16(i)) | * | Minus | ** | = | X $ = | | OR | X $ = | |
| | Independent (37 CFR 1.16(h)) | * | Minus | *** | = | X $ = | | OR | X $ = | |
| | ☐ Application Size Fee (37 CFR 1.16(s)) | | | | | | | | | |
| | ☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j)) | | | | | | | OR | | |
| | | | | | | TOTAL ADD'L FEE | | OR | TOTAL ADD'L FEE | |

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.

** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20".

*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3".

The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

Legal Instrument Examiner:
/EAON SMALLWOOD/

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**

*If you need assistance in completing the form, call 1-800-PTO-9199 and select option 2.*

| Form 1449 (Modified) | Atty Docket No. | REBLP003/VOX-P003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 1 of 1 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 1. | | | | | | |
| | 2. | | | | | | |
| | 3. | | | | | | |
| | 4. | | | | | | |

**Foreign Patent or Published Foreign Patent Application**

| Examiner Initial | No. | Document No. | Publication Date | Country or Patent Office | Class | Sub-Class | Translation Yes | Translation No |
|---|---|---|---|---|---|---|---|---|
| | 5. | | | | | | | |
| | 6. | | | | | | | |
| | 7. | | | | | | | |
| | 8. | | | | | | | |

**Other Documents**

| Examiner Initial | No. | Author, Title, Place (e.g. Journal) of Publication, Date |
|---|---|---|
| | 9. | Office Action in EP 08747806.1, dated April 27, 2010. |
| | 10. | |
| | 11. | |
| | 12. | |
| | 13. | |
| | 14. | |
| | 15. | |
| | 16. | |
| | 17. | |
| | 18. | |
| | 19. | |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 7811024 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 14-JUN-2010 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 19:14:56 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Transmittal Letter | IDS_Regular_Transmittal3.pdf | 23928<br>18ffcf4••d2157caa9•4441ef2•00e215815b50495 | no | 2 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |

| | | | | | |
|---|---|---|---|---|---|
| 2 | Information Disclosure Statement (IDS) Filed (SB/08) | 1449_Form_IDS3.pdf | 25029<br>04dfff7c682e025ea550e66213f993f5b29364c4 | no | 1 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| This is not an USPTO supplied IDS fillable form | | | | | |
| 3 | NPL Documents | REBLP003AEP_OA_4_27_10.pdf | 73952<br>22023aa31e84c1e14576df66c7f2bfa76c271202 | no | 3 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| | | **Total Files Size (in bytes):** | 122909 | | |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**

**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**

**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**

**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et. al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: KUNTZ, CURTIS A |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on June 14, 2010.

Signed: /Laura M. Dean/
Laura M. Dean

# INFORMATION DISCLOSURE STATEMENT
# 37 CFR §§1.56 AND 1.97(b)

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The reference(s) identified in the attached PTO Form 1449 (copies of non-U.S. references are attached) may be material to examination of the above-identified patent application. Applicants identify the reference(s) in compliance with their duty of disclosure pursuant to 37 CFR §§1.56 and 1.97. The Examiner is requested to make the identified reference(s) of official record in this application.

This Information Disclosure Statement is not to be construed as a representation that a search has been made, that additional information material to the examination of this application does not exist, or that the identified reference(s) indeed constitute prior art.

This Information Disclosure Statement is: (i) filed within three (3) months of the filing date of the above-referenced application, (ii) believed to be filed before the mailing date of a first Office Action on the merits, or (iii) believed to be filed before the mailing of a first Office Action after the filing of a Request for Continued Examination under §1.114. Accordingly, it is believed that no fees are due in connection with the filing of this Information Disclosure

Statement. However, if it is determined that any fees are due, the Commissioner is hereby authorized to charge such fees to Deposit Account 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

| Form 1449 (Modified) | Atty Docket No. | REBLP003/VOX-P003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS, et al. |
| | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 1 of 1 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 1. | | | | | | |
| | 2. | | | | | | |
| | 3. | | | | | | |
| | 4. | | | | | | |

**Foreign Patent or Published Foreign Patent Application**

| Examiner Initial | No. | Document No. | Publication Date | Country or Patent Office | Class | Sub-Class | Translation Yes | Translation No |
|---|---|---|---|---|---|---|---|---|
| | 5. | | | | | | | |
| | 6. | | | | | | | |
| | 7. | | | | | | | |
| | 8. | | | | | | | |

**Other Documents**

| Examiner Initial | No. | Author, Title, Place (e.g. Journal) of Publication, Date |
|---|---|---|
| | 9. | Office Action in EP 08769254.7, dated July 12, 2010. |
| | 10. | |
| | 11. | |
| | 12. | |
| | 13. | |
| | 14. | |
| | 15. | |
| | 16. | |
| | 17. | |
| | 18. | |
| | 19. | |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 8359098 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/Laura Dean |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 04-SEP-2010 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 20:02:41 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Transmittal Letter | IDS_Regular_Transmittal4.pdf | 23804<br>763422bb3a6fcd23cce703f0f5f6f6d89be1Ce2a | no | 2 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |

| | | | | | |
|---|---|---|---|---|---|
| 2 | Information Disclosure Statement (IDS) Filed (SB/08) | 1449_Form_IDS4.pdf | 25076<br>e7290fc6424bb6b722ee2449ef651[redacted]8d9<br>36142 | no | 1 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| This is not an USPTO supplied IDS fillable form | | | | | |
| 3 | NPL Documents | REBLP003EP_OA_7_12_10.pdf | 282997<br>29519bf3[redacted]b8fe0e48eb425bc17aa46a1cf20<br>ca7e | no | 6 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| | | **Total Files Size (in bytes):** | 331877 | | |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**
**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**
**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**
**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et. al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: RAMAKRISHNAIAH, MELUR |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on September 4, 2010.

Signed: /Laura M. Dean/
Laura M. Dean

# INFORMATION DISCLOSURE STATEMENT
# 37 CFR §§1.56 AND 1.97(b)

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The reference(s) identified in the attached PTO Form 1449 (copies of non-U.S. references are attached) may be material to examination of the above-identified patent application. Applicants identify the reference(s) in compliance with their duty of disclosure pursuant to 37 CFR §§1.56 and 1.97. The Examiner is requested to make the identified reference(s) of official record in this application.

This Information Disclosure Statement is not to be construed as a representation that a search has been made, that additional information material to the examination of this application does not exist, or that the identified reference(s) indeed constitute prior art.

This Information Disclosure Statement is: (i) filed within three (3) months of the filing date of the above-referenced application, (ii) believed to be filed before the mailing date of a first Office Action on the merits, or (iii) believed to be filed before the mailing of a first Office Action after the filing of a Request for Continued Examination under §1.114. Accordingly, it is believed that no fees are due in connection with the filing of this Information Disclosure

Statement. However, if it is determined that any fees are due, the Commissioner is hereby authorized to charge such fees to Deposit Account 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 1 of 2 |

**U.S. Patent Documents**

| Examiner Initial | No. | Patent No. | Date | Patentee | Class | Sub-class | Filing Date |
|---|---|---|---|---|---|---|---|
| | 1. | 2002/0091848 | 07/11/02 | Agresta et al. | | | 09/05/01 |
| | 2. | 2002/0146110 | 10/10/02 | Fromm | | | 04/06/01 |
| | 3. | 2004/0044783 | 03/04/04 | Nordberg | | | 10/24/02 |
| | 4. | 2006/0221995 | 10/05/06 | Berkman | | | 04/04/05 |
| | 5. | 2006/0248149 | 11/02/06 | Kraft et al. | | | 04/28/05 |
| | 6. | 2007/0021131 | 01/25/07 | Laumen et al. | | | 02/16/05 |
| | 7. | 2007/0159321 | 07/12/07 | Ogata et al. | | | 12/21/06 |
| | 8. | 2007/0263575 | 11/15/07 | Choe | | | 11/16/06 |
| | 9. | 2007/0268887 | 11/22/07 | Schwartz | | | 05/19/06 |
| | 10. | 2008/0095173 | 04/24/08 | Bugenhagen | | | 10/19/06 |
| | 11. | 2008/0115087 | 05/15/08 | Rollin et al. | | | 11/13/06 |
| | 12. | 2008/0147910 | 06/19/08 | Hibbard et al. | | | 09/29/06 |
| | 13. | 2009/0003339 | 01/01/09 | Katis et al. | | | 02/15/08 |
| | 14. | 2009/0003340 | 01/01/09 | Katis et al. | | | 02/26/08 |
| | 15. | 2009/0003547 | 01/01/09 | Katis et al. | | | 05/16/08 |
| | 16. | 2009/0003558 | 01/01/09 | Katis et al. | | | 02/08/08 |
| | 17. | 2009/0298485 | 12/03/09 | Lee et al. | | | 08/17/09 |
| | 18. | 2010/0215158 | 08/26/10 | Katis et al. | | | 05/06/10 |
| | 19. | 2010/0217822 | 08/26/10 | Katis et al. | | | 05/06/10 |
| | 20. | 5,651,054 | 07/22/97 | Dunn, et al. | | | 04/13/95 |
| | 21. | 6,389,473 | 05/14/02 | Carmel, et al. | | | 03/24/99 |

**Foreign Patent or Published Foreign Patent Application**

| Examiner Initial | No. | Document No. | Publication Date | Country or Patent Office | Class | Sub-Class | Translation Yes | Translation No |
|---|---|---|---|---|---|---|---|---|
| | 22. | EP 1379061 | 01/07/04 | EPO | | | | |
| | 23. | EP 1715648 | 10/25/06 | EPO | | | | |
| | 24. | WO 2006/096557 | 09/14/06 | PCT | | | | |
| | 25. | WO 2007/081929 | 07/19/07 | PCT | | | | |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

| Form 1449 (Modified) | Atty Docket No. | REBLP003 |
|---|---|---|
| | Application No.: | 12/037,256 |
| Information Disclosure Statement By Applicant | Applicant | KATIS et al. |
| | Filing Date | February 26, 2008 |
| | Group | 2614 |
| (Use Several Sheets if Necessary) | | Page 2 of 2 |

**Other Documents**

| Examiner Initial | No. | Author, Title, Place (e.g. Journal) of Publication, Date |
|---|---|---|
| | 26. | Office Action in U.S. 12/042,987, dated November 24, 2010. |
| | 27. | Office Action in U.S. 12/102,722, dated December 8, 2010. |
| | 28. | Office Action in U.S. 12/103,436, dated December 8, 2010. |
| | 29. | Office Action in EP 08747806.1, dated November 18, 2010. |

| Examiner | Date Considered |
|---|---|
| | |

Examiner: Initial citation considered. Draw line through citation if not in conformance and not considered. Include copy of this form with next communication to applicant.

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 379 061 A2

# (12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
**07.01.2004 Bulletin 2004/02**

(51) Int Cl.[7]: **H04M 1/65**, H04M 3/533

(21) Application number: **03253638.5**

(22) Date of filing: **09.06.2003**

(84) Designated Contracting States:
**AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IT LI LU MC NL PT RO SE SI SK TR**
Designated Extension States:
**AL LT LV MK**

(30) Priority: **10.06.2002 US 386709 P**

(71) Applicant: **Research In Motion Limited**
**Waterloo, Ontario N2L 3W8 (CA)**

(72) Inventors:
- **Vander Veen, Raymond P.**
  **Waterloo, Ontario N2V 2M6 (CA)**
- **Lazaridis, Mihal**
  **Waterloo, Ontario N2T 2K1 (CA)**

(74) Representative: **Walaski, Jan Filip et al**
**Venner, Shipley & Co,**
**20 Little Britain**
**London EC1A 7DH (GB)**

## (54) Voicemail user interface methods and apparatus for mobile communication devices

(57) Voicemail notification messaging for mobile communication devices is disclosed. Voicemail notification messages corresponding to voicemail messages are received through a wireless receiver of the mobile device and voicemail message header lines corresponding to them are displayed in a message list. The mobile device (202) receives the voicemail notification messages from several different voicemail systems (302) for consolidating voicemail message processing using a single common user interface. Preferably, the mobile device provides a graphical user interface (GUI) for the end-user with visual objects corresponding to PLAY, REWIND, FAST FORWARD, SKIP BACK, SKIP FORWARD, as examples, which provides for transparency of voicemail system-specific commands. Advantageously, each voicemail notification message includes voicemail message summary information, such as a calling party identifier, a message time stamp, and message length, for use in providing voicemail message header information in the message list. In addition, each voicemail notification message includes voicemail message retrieval/processing information, such as a voicemail access telephone number, a voicemail message identification number, and voicemail message processing commands, for use in providing the system interface transparency.

EP 1 379 061 A2


Printed by Jouve, 75001 PARIS (FR)

## Description

### BACKGROUND

#### Field of the Invention

**[0001]** The present invention relates generally to mobile communication devices and the associated communication networks within which they operate, and more particularly to the receiving and sending of voicemail notification messages which include voicemail message retrieval/processing information for use by mobile communication devices.

#### Description of the Related Art

**[0002]** "Voicemail" is a common feature for practically all voice-based communication products. As examples, home telephone systems now offer voicemail features built right into telephones, telephone companies offer voicemail services in their phone networks, corporations provide voicemail for every desk with integration with e-mail, and wireless carriers offer voicemail packages in connection with their cellular telephones. The average professional may have two or three different voicemail systems that must be checked for new voicemail messages from the home, the office, or cellular telephone. Each voicemail system typically has a different password access and different command codes for the same voicemail functions (e.g. PLAY, REWIND, SAVE, and DELETE).

**[0003]** The result of the above is that the end user must check several different voicemail systems for voicemail while away and remember each set of voicemail command codes. One way to handle this problem is to manually forward all phone calls (i.e. via call forwarding) from all devices to a common unified messaging service (UMS). Here, the end user can check all messages at once from a single voicemail location. However, this solution is prone to problems since the end user may forget to forward phone calls as necessary, may be too busy traveling to regularly check for voicemail messages, and may forget passwords needed to access the voicemail systems.

**[0004]** Accordingly, there is a strong need to provide unified methods and apparatus for handling and processing voicemail messages to reduce the problems created by today's use of multiple voicemail systems. There is also a strong need to simplify the user interface for retrieving and processing voicemail messages from several different voicemail systems.

### SUMMARY

**[0005]** Unique voicemail notification messaging for mobile communication devices is described herein. Voicemail notification messages corresponding to voicemail messages are received through a wireless receiver of the mobile device and voicemail message header lines corresponding to them are displayed in a message list. The mobile device receives the voicemail notification messages from several different voicemail systems for consolidating voicemail retrieval and processing using a single common user interface. Preferably, the mobile device provides a graphical user interface (GUI) for the end-user with visual objects corresponding to PLAY, REWIND, FAST FORWARD, SKIP BACK, SKIP FORWARD, as examples, which provides for transparency of voicemail system-specific commands.

**[0006]** Advantageously, each voicemail notification message includes voicemail message summary information, such as a calling party identifier, a message time stamp, and message length, for use in providing voicemail message header information in the message list. In addition, each voicemail notification message includes voicemail message retrieval/processing information, such as a voicemail access telephone number, a voicemail message identification number, and voicemail message processing commands, for use in providing the system interface transparency.

### BRIEF DESCRIPTION OF THE DRAWINGS

**[0007]** Embodiments of present invention will now be described by way of example with reference to attached figures, wherein:

FIG. 1 is a block diagram which illustrates pertinent components of a mobile communication device which communicates within a wireless communication network;
FIG. 2 is a more detailed diagram of a preferred mobile communication device of FIG. 1;
FIG. 3 is a diagram of communication network components which may be used in connection with the methods described herein;
FIG. 4 is an example of voicemail and database information which may be used for that described in the present application;
FIG. 5 is another example of voicemail and database information which may be used for that described in the

present application;
FIG. 6 is yet another example of voicemail and database information which may be used for that described in the present application;
FIG. 7 is a message format which may be used for communication of voicemail notification messages having voicemail notification payloads (VNPs);
FIG. 8 is a flowchart of a general method of sending voicemail notification information from a voicemail system to a mobile communication device;
FIG. 9 is a flowchart of a general method of receiving and processing voicemail notification information by a mobile communication device;
FIG. 10 is a flowchart which describes an illustrative overview of methods of the present application; and
FIGs. 11-16 are illustrations of a visual display of the mobile communication device, showing voicemail message data and a graphical user interface (GUI) for retrieving and processing voicemail messages.

## DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

**[0008]** Voicemail notification messaging for mobile communication devices is described herein. Voicemail notification messages corresponding to voicemail messages are received through a wireless receiver of the mobile device and voicemail message header lines corresponding to them are displayed in a message list. The mobile device receives the voicemail notification messages from several different voicemail systems for consolidating voicemail retrieval and processing using a single common user interface. Preferably, the mobile device provides a graphical user interface (GUI) for the end-user with visual objects corresponding to PLAY, REWIND, FAST FORWARD, SKIP BACK, SKIP FORWARD, as examples, which provides for transparency of voicemail system-specific commands. Advantageously, each voicemail notification message includes voicemail message summary information, such as a calling party identifier, a message time stamp, and message length, for use in providing voicemail message header information in the message list. In addition, each voicemail notification message includes voicemail message retrieval/processing information, such as a voicemail access telephone number, a voicemail message identification number, and voicemail message processing commands, for use in providing the system interface transparency.

**[0009]** FIG. 1 is a block diagram of a communication system 100 which includes a mobile station 102 which communicates through a wireless communication network 104. In the embodiment of FIG. 1, wireless network 104 is configured in accordance with General Packet Radio Service (GPRS) and a Global Systems for Mobile (GSM) technologies; however, any suitable type of network communication protocols may be utilized. For example, the network may be based on code division multiple access (CDMA) or other suitable technologies. As another example, the network may be based on an Integrated Dispatch Enhanced Network (iDEN) which is a high-capacity digital trunked radio system providing integrated voice and data services.

**[0010]** Mobile station 102, which is one type of mobile communication device, preferably includes a visual display 112, a keyboard 114, and perhaps one or more auxiliary user interfaces (UI) 116, each of which are coupled to a controller 106. Controller 106 is also coupled to radio frequency (RF) transceiver circuitry 108 and an antenna 110. Typically, controller 106 is embodied as a central processing unit (CPU) which runs operating system software in a memory component (not shown). Controller 106 will normally control overall operation of mobile station 102, whereas signal processing operations associated with communication functions are typically performed in RF transceiver circuitry 108. Controller 106 interfaces with device display 112 to display received information, stored information, user inputs, and the like. Keyboard 114, which may be a telephone type keypad or full alphanumeric keyboard, is normally provided for entering data for storage in mobile station 102, information for transmission to network 104, a telephone number to place a telephone call, commands to be executed on mobile station 102, and possibly other or different user inputs.

**[0011]** Mobile station 102 sends communication signals to and receives communication signals from network 104 over a wireless link via antenna 110. RF transceiver circuitry 108 performs functions similar to those of station 118 and base station controller 120, including for example modulation/demodulation and possibly encoding/decoding and encryption/decryption. It is also contemplated that RF transceiver circuitry 108 may perform certain functions in addition to those performed by base station controller 120. It will be apparent to those skilled in art that RF transceiver circuitry 108 will be adapted to particular wireless network or networks in which mobile station 102 is intended to operate.

**[0012]** Mobile station 102 includes a battery interface 134 for receiving one or more rechargeable batteries 132. Battery 132 provides electrical power to electrical circuitry in mobile station 102, and battery interface 132 provides for a mechanical and electrical connection for battery 132. Battery interface 132 is coupled to a regulator 136 which regulates power to the device. When mobile station 102 is fully operational, an RF transmitter of RF transceiver circuitry 108 is typically keyed or turned on only when it is sending to network, and is otherwise turned off to conserve resources. Similarly, an RF receiver of RF transceiver circuitry 108 is typically periodically turned off to conserve power until it is needed to receive signals or information (if at all) during designated time periods.

**[0013]** Mobile station 102 operates using a Subscriber Identity Module (SIM) 140 which is connected to or inserted in mobile station 102 at a SIM interface 142. SIM 140 is one type of a conventional "smart card" used to identify an end user (or subscriber) of mobile station 102 and to personalize the device, among other things. Without SIM 140, the mobile station terminal is not fully operational for communication through wireless network 104. By inserting SIM 140 into mobile station 102, an end user can have access to any and all of his/her subscribed services. SIM 140 generally includes a processor and memory for storing information. Since SIM 140 is coupled to SIM interface 142, it is coupled to controller 106 through communication lines 144. In order to identify the subscriber, SIM 140 contains some user parameters such as an International Mobile Subscriber Identity (IMSI). An advantage of using SIM 140 is that end users are not necessarily bound by any single physical mobile station. SIM 140 may store additional user information for the mobile station as well, including datebook (or calendar) information and recent call information.

**[0014]** Mobile station 102 may consist of a single unit, such as a data communication device, a cellular telephone, a multiple-function communication device with data and voice communication capabilities, a personal digital assistant (PDA) enabled for wireless communication, or a computer incorporating an internal modem. Alternatively, mobile station 102 may be a multiple-module unit comprising a plurality of separate components, including but in no way limited to a computer or other device connected to a wireless modem. In particular, for example, in the mobile station block diagram of FIG. 1, RF transceiver circuitry 108 and antenna 110 may be implemented as a radio modem unit that may be inserted into a port on a laptop computer. In this case, the laptop computer would include display 112, keyboard 114, one or more auxiliary UIs 116, and controller 106 embodied as the computer's CPU. It is also contemplated that a computer or other equipment not normally capable of wireless communication may be adapted to connect to and effectively assume control of RF transceiver circuitry 108 and antenna 110 of a single-unit device such as one of those described above. Such a mobile station 102 may have a more particular implementation as described later in relation to mobile station 402 of FIG. 2.

**[0015]** Mobile station 102 communicates in and through wireless communication network 104. In the embodiment of FIG. 1, wireless network 104 is configured in accordance with General Packet Radio Service (GPRS) and a Global Systems for Mobile (GSM) technologies. Wireless network 104 includes a base station controller (BSC) 120 with an associated tower station 118, a Mobile Switching Center (MSC) 122, a Home Location Register (HLR) 132, a Serving General Packet Radio Service (GPRS) Support Node (SGSN) 126, and a Gateway GPRS Support Node (GGSN) 128. MSC 122 is coupled to BSC 120 and to a landline network, such as a Public Switched Telephone Network (PSTN) 124. SGSN 126 is coupled to BSC 120 and to GGSN 128, which is in turn coupled to a public or private data network 130 (such as the Internet). HLR 132 is coupled to MSC 122, SGSN 126, and GGSN 128.

**[0016]** Station 118 is a fixed transceiver station, and station 118 and BSC 120 are together referred to herein as the fixed transceiver equipment. The fixed transceiver equipment provides wireless network coverage for a particular coverage area commonly referred to as a "cell". The fixed transceiver equipment transmits communication signals to and receives communication signals from mobile stations within its cell via station 118. The fixed transceiver equipment normally performs such functions as modulation and possibly encoding and/or encryption of signals to be transmitted to the mobile station in accordance with particular, usually predetermined, communication protocols and parameters, under control of its controller. The fixed transceiver equipment similarly demodulates and possibly decodes and decrypts, if necessary, any communication signals received from mobile station 102 within its cell. Communication protocols and parameters may vary between different networks. For example, one network may employ a different modulation scheme and operate at different frequencies than other networks.

**[0017]** The wireless link shown in communication system 100 of FIG. 1 represents one or more different channels, typically different radio frequency (RF) channels, and associated protocols used between wireless network 104 and mobile station 102. Those skilled in art will appreciate that a wireless network in actual practice may include hundreds of cells, each served by a station 118 (i.e. or station sector), depending upon desired overall expanse of network coverage. All pertinent components may be connected by multiple switches and routers (not shown), controlled by multiple network controllers.

**[0018]** For all mobile station's 102 registered with a network operator, permanent data (such as mobile station 102 user's profile) as well as temporary data (such as mobile station's 102 current location) are stored in HLR 132. In case of a voice call to mobile station 102, HLR 132 is queried to determine the current location of mobile station 102. A Visitor Location Register (VLR) of MSC 122 is responsible for a group of location areas and stores the data of those mobile stations that are currently in its area of responsibility. This includes parts of the permanent mobile station data that have been transmitted from HLR 132 to the VLR for faster access. However, the VLR of MSC 122 may also assign and store local data, such as temporary identifications. Optionally, the VLR of MSC 122 can be enhanced for more efficient co-ordination of GPRS and non-GPRS services and functionality (e.g. paging for circuit-switched calls which can be performed more efficiently via SGSN 126, and combined GPRS and non-GPRS location updates).

**[0019]** Serving GPRS Support Node (SGSN) 126 is at the same hierarchical level as MSC 122 and keeps track of the individual locations of mobile stations. SGSN 126 also performs security functions and access control. Gateway GPRS Support Node (GGSN) 128 provides interworking with external packet-switched networks and is connected with

SGSNs (such as SGSN 126) via an IP-based GPRS backbone network. SGSN 126 performs authentication and cipher setting procedures based on the same algorithms, keys, and criteria as in existing GSM. In conventional operation, cell selection may be performed autonomously by mobile station 102 or by the fixed transceiver equipment instructing mobile station 102 to select a particular cell. Mobile station 102 informs wireless network 104 when it reselects another cell or group of cells, known as a routing area.

**[0020]** In order to access GPRS services, mobile station 102 first makes its presence known to wireless network 104 by performing what is known as a GPRS "attach". This operation establishes a logical link between mobile station 102 and SGSN 126 and makes mobile station 102 available to receive, for example, pages via SGSN, notifications of incoming data, or SMS messages over GPRS. In order to send and receive GPRS data, mobile station 102 assists in activating the packet data address that it wants to use. This operation makes mobile station 102 known to GGSN 128; interworking with external data networks can thereafter commence. User data may be transferred transparently between mobile station 102 and the external data networks using, for example, encapsulation and tunneling. Data packets are equipped with GPRS-specific protocol information and transferred between mobile station 102 and GGSN 128.

**[0021]** FIG. 2 is a detailed block diagram of a preferred mobile station 202. Mobile station 202 is preferably a two-way communication device having at least voice and advanced data communication capabilities, including the capability to communicate with other computer systems. Depending on the functionality provided by mobile station 202, it may be referred to as a data messaging device, a two-way pager, a cellular telephone with data messaging capabilities, a wireless Internet appliance, or a data communication device (with or without telephony capabilities). Mobile station 202 may communicate with any one of a plurality of fixed transceiver stations 200 within its geographic coverage area.

**[0022]** Mobile station 202 will normally incorporate a communication subsystem 211, which includes a receiver 212, a transmitter 214, and associated components, such as one or more (preferably embedded or internal) antenna elements 216 and 218, local oscillators (LOs) 213, and a processing module such as a digital signal processor (DSP) 220. Communication subsystem 211 is analogous to RF transceiver circuitry 108 and antenna 110 shown in FIG. 1. As will be apparent to those skilled in field of communications, particular design of communication subsystem 211 depends on the communication network in which mobile station 202 is intended to operate.

**[0023]** Mobile station 202 may send and receive communication signals over the network after required network registration or activation procedures have been completed. Signals received by antenna 216 through the network are input to receiver 212, which may perform such common receiver functions as signal amplification, frequency down conversion, filtering, channel selection, and like, and in example shown in FIG. 2, analog-to-digital (A/D) conversion. A/D conversion of a received signal allows more complex communication functions such as demodulation and decoding to be performed in DSP 220. In a similar manner, signals to be transmitted are processed, including modulation and encoding, for example, by DSP 220. These DSP-processed signals are input to transmitter 214 for digital-to-analog (D/A) conversion, frequency up conversion, filtering, amplification and transmission over communication network via antenna 218. DSP 220 not only processes communication signals, but also provides for receiver and transmitter control. For example, the gains applied to communication signals in receiver 212 and transmitter 214 may be adaptively controlled through automatic gain control algorithms implemented in DSP 220.

**[0024]** Network access is associated with a subscriber or user of mobile station 202, and therefore mobile station 202 requires a Subscriber Identity Module or "SIM" card 262 to be inserted in a SIM interface 264 in order to operate in the network. SIM 262 includes those features described in relation to FIG. 1. Mobile station 202 is a battery-powered device so it also includes a battery interface 254 for receiving one or more rechargeable batteries 256. Such a battery 256 provides electrical power to most if not all electrical circuitry in mobile station 202, and battery interface 254 provides for a mechanical and electrical connection for it. The battery interface 254 is coupled to a regulator (not shown) which provides power V+ to all of the circuitry.

**[0025]** Mobile station 202 includes a microprocessor 238 (which is one implementation of controller 106 of FIG. 1) which controls overall operation of mobile station 202. Communication functions, including at least data and voice communications, are performed through communication subsystem 211. Microprocessor 238 also interacts with additional device subsystems such as a display 222, a flash memory 224, a random access memory (RAM) 226, auxiliary input/output (I/O) subsystems 228, a serial port 230, a keyboard 232, a speaker 234, a microphone 236, a short-range communications subsystem 240, and any other device subsystems generally designated at 242. Some of the subsystems shown in FIG. 2 perform communication-related functions, whereas other subsystems may provide "resident" or on-device functions. Notably, some subsystems, such as keyboard 232 and display 222, for example, may be used for both communication-related functions, such as entering a text message for transmission over a communication network, and device-resident functions such as a calculator or task list. Operating system software used by microprocessor 238 is preferably stored in a persistent store such as flash memory 224, which may alternatively be a read-only memory (ROM) or similar storage element (not shown). Those skilled in the art will appreciate that the operating system, specific device applications, or parts thereof, may be temporarily loaded into a volatile store such as RAM 226.

**[0026]** Microprocessor 238, in addition to its operating system functions, preferably enables execution of software applications on mobile station 202. A predetermined set of applications which control basic device operations, including

at least data and voice communication applications, will normally be installed on mobile station 202 during its manufacture. A preferred application which is loaded onto mobile station 202 may be a personal information manager (PIM) application having the ability to organize and manage data items relating to e-mail messages and voicemail messages, as well as calendar data. Naturally, one or more memory stores are available on mobile station 202 and SIM 256 to facilitate storage of PIM data items and other information.

**[0027]** The PIM application preferably has the ability to send and receive data items via the wireless network. In a preferred embodiment, PIM data items are seamlessly integrated, synchronized, and updated via the wireless network, with the mobile station user's corresponding data items stored and/or associated with a host computer system thereby creating a mirrored host computer on mobile station 202 with respect to such items. This is especially advantageous where the host computer system is the mobile station user's office computer system. Additional applications may also be loaded onto mobile station 202 through network, an auxiliary I/O subsystem 228, serial port 230, short-range communications subsystem 240, or any other suitable subsystem 242, and installed by a user in RAM 226 or preferably a non-volatile store (not shown) for execution by microprocessor 238.

**[0028]** In a data communication mode, a received signal such as a text message (e.g. a short message service or SMS message), an e-mail message, or web page download will be processed by communication subsystem 211 and input to microprocessor 238. Microprocessor 238 will preferably further process the signal for output to display 222 or alternatively to auxiliary I/O device 228. A user of mobile station 202 may also compose data items, such as e-mail messages, for example, using keyboard 232 in conjunction with display 222 and possibly auxiliary I/O device 228. Keyboard 232 is preferably a complete alphanumeric keyboard and/or telephone-type keypad. These composed items may be transmitted over a communication network through communication subsystem 211.

**[0029]** For voice communications, the overall operation of mobile station 202 is substantially similar, except that the received signals would be output to speaker 234 and signals for transmission would be generated by microphone 236. Alternative voice or audio I/O subsystems, such as a voice message recording subsystem, may also be implemented on mobile station 202. Although voice or audio signal output is preferably accomplished primarily through speaker 234, display 222 may also be used to provide an indication of the identity of a calling party, duration of a voice call, or other voice call related information, as some examples.

**[0030]** Serial port 230 in FIG. 2 is normally implemented in a personal digital assistant (PDA)-type communication device for which synchronization with a user's desktop computer is a desirable, albeit optional, component. Serial port 230 enables a user to set preferences through an external device or software application and extends the capabilities of mobile station 202 by providing for information or software downloads to mobile station 202 other than through a wireless communication network. The alternate download path may, for example, be used to load an encryption key onto mobile station 202 through a direct and thus reliable and trusted connection to thereby provide secure device communication.

**[0031]** Short-range communications subsystem 240 of FIG. 2 is an additional optional component which provides for communication between mobile station 202 and different systems or devices, which need not necessarily be similar devices. For example, subsystem 240 may include an infrared device and associated circuits and components, or a Bluetooth™ communication module to provide for communication with similarly-enabled systems and devices. Bluetooth™ is a registered trademark of Bluetooth SIG, Inc.

**[0032]** FIG. 3 is a diagram of communication network components for use in describing a general overview of the techniques of the present application. An external source 300, such as a telephone or mobile phone, places a call 302. Call 302 may be routed to a corporate telephone 308 or to another service provider's mobile phone 324. When call 302 is intended for corporate telephone 308, call 302 is routed through a corporate phone service 304 and delivered 306 to corporate telephone 308 which is answered by the user. If the user is unable to answer corporate phone 308, however, the call is sent back 306 to corporate phone service 304 where the caller leaves a voicemail message 310. Voicemail message 310 is left at a unified messaging notification system 312 where it is stored. Without initiation from mobile device 322, system 312 immediately thereafter sends a voicemail notification message 314 via the Internet 318 and a wireless network 320 to the user's mobile device 322. Voicemail notification message 314 may be in the form of an e-mail message or a short message service (SMS) message. Based on voicemail summary information provided in voicemail notification message 314, voicemail message header information associated with voicemail message 310 is displayed in a visual display of mobile device 322. The header information may include, but is not limited to, calling party identification information, time stamp information, and time duration information.

**[0033]** When the user chooses to retrieve voicemail message 310, the user selects a visual object or switch associated with a "PLAY" function in connection with the header message. In response, mobile device 322 places a call 346 to unified messaging notification system 312 and sends one or more commands to play the voicemail message 310 at the mobile device 322. Preferably, voicemail notification message 314 includes not only the information used to provided the header information displayed in the visual display, but also voicemail message retrieval information used to access and process voicemail message 310 from system 312. This information may include, but is not limited to, a voicemail access telephone number, a message identification number, a voicemail system command to select

and/or play the voicemail, and other voicemail system commands such as those relating to rewind, fast forward, skip back, skip forward, save, and delete, as examples. During or after voicemail message 310 is played, the user may select from a plurality of switches or visual objects in the display that are associated with "REWIND", "FAST FORWARD", "SKIP BACK", "SKIP FORWARD", "SAVE", or "DELETE", as examples. Mobile device 322 causes the stored voicemail system command corresponding to the user selected function to be sent to system 312 for executing the corresponding function at system 312 for processing voicemail message 310.

**[0034]** On the other hand, if call 302 is placed to a mobile phone 324 associated with the same end user, the call is sent to the mobile phone carrier's mailbox 328 where the caller leaves a voicemail message 330 if the user does not answer. The voicemail message 330 is sent to a unified messaging notification system 332 for the mobile phone's carrier where it is stored. Without initiation from mobile device 322, system 332 sends a voicemail notification message 334 via the Internet 318 and wireless network 320 to the user's mobile device 322. Voicemail notification message 334 may be in the form of an e-mail message or a short message service (SMS) message. Based on voicemail summary information provided in voicemail notification message 334, voicemail message header information associated with voicemail message 330 is displayed in a visual display of mobile device 322. The header information may include, but is not limited to, calling party identification information, time stamp information, and time duration information.

**[0035]** When the user chooses to retrieve voicemail message 330, the user selects a switch or visual object in the visual display associated with a "PLAY" function for this message. In response, mobile device 322 places a call 346 to the unified messaging system 332 and sends one or more commands to play the voicemail 330 at the mobile device 322. Preferably, voicemail notification message 334 includes not only the information used to provided the header information displayed in the visual display, but also voicemail message retrieval information used to access and process voicemail message 330 from system 332. This information may include, but is not limited to, a voicemail access telephone number, a message identification number, a voicemail system command to select and/or play the voicemail, and other voicemail system commands such as those relating to rewind, fast forward, skip back, skip forward, save, and delete, as examples. During or after voicemail message 330 is played, the user may select from a plurality of switches or visual objects in the display that are associated with "REWIND", "FAST FORWARD", "SKIP BACK", "SKIP FORWARD", "SAVE", or "DELETE", as examples. Mobile device 322 causes the stored voicemail system command corresponding to the user selected function to be sent to system 332 for executing the corresponding function at system 332 for processing voicemail message 330. As apparent from the above in FIG. 3, transparency of voicemail system-specific commands is provided in connection with use of several different voicemail systems.

**[0036]** FIG. 4 is one example illustration of voicemail processing data which may stored in a mobile device. A software application 401 on a mobile device 400 receives a voicemail notification message which carries a voicemail notification payload (VNP) 404. As initially received, VNP 404 may be embodied in a short message service (SMS) message or, alternatively, in an attachment of an e-mail message. VNP 404 may include information which is made visible to the user in mobile device's display 402. This information, which may be referred to as voicemail message summary information, may include information such as the length of the message, the caller's identification, the caller's phone number, and the time and date of the voicemail. VNP 404 may also include information not made visible to the user but used by mobile device 400 to retrieve, play, and further process voicemail messages from a voicemail system. This information, which may be referred to as voicemail message retrieval information, may include information such as the identification of the service provider's mailbox (mailbox ID), the protocol that the service provider uses (protocol ID), the version of the protocol used, information about the length of DTMF tones required by the protocol, and the number that the device must call to retrieve the voicemail message.

**[0037]** VNP 404 is associated with a database 406 on mobile device 400 with that sets out different command sets for different voicemail system protocols. That is, for each different protocol ID, database 406 associates a set of voicemail functions with their appropriate DTMF tone commands 408 used by the service provider to access and process voicemail messages. When a user selects a voicemail processing function (e.g. FAST FORWARD) at the user interface, mobile device 400 causes the associated DTMF tone command for that function to be sent. Database 406 may be populated with this information from a service provider when mobile device 400 makes an initial call to the service provider. Alternatively, a user may also populate database 406 with this information. Database 406 is further associated with another database (not shown) on mobile device 400 that consists of the user passwords for each unified messaging system from which mobile device 400 receives voicemail notification messages.

**[0038]** FIG. 5 shows a different example of voicemail processing data on a mobile device 500. Again, a software application 501 on a mobile device 500 receives a voicemail notification message which carries a voicemail notification payload (VNP) 504. As initially received, VNP 504 may be embodied in an SMS message or, alternatively, in an attachment of an e-mail message. As in FIG. 4, VNP 504 may include information which is made visible to the user in mobile device's display 502. This information, which may be referred to as voicemail message summary information, may include information such as the length of the message, the calling party's identification, the caller's phone number, and the time and date of the voicemail. VNP 504 includes other information as shown, including bit flags to determined whether a password is required or not, a mailbox ID, information about the required lengths of DTMF tones, a voicemail

system access number, etc. VNP 504 also includes all the DTMF tone commands necessary for mobile device 500 to play and process voicemail messages, instead of the prestored protocol information described in relation to FIG. 4. This difference in VNPs means that mobile device 500 no longer requires the different commands for each protocol to be prestored in the database, but rather receives such commands in each voicemail notification message. Mobile device 500 only needs a database 507 of passwords 509 associated with each mailbox ID which the device references when a given password 509 is required by the unified messaging system for authentication.

**[0039]** FIG. 6 shows yet another example of voicemail processing data on a mobile device 600. Again, a software application 601 on a mobile device 600 receives a voicemail notification message which carries a voicemail notification payload (VNP) 604. As initially received, VNP 604 may be embodied in an SMS message or, alternatively, in an attachment of an e-mail message. As in FIGs. 4 and 5, VNP 604 may include information which is made visible to the user in mobile device's display 602. This information, which may be referred to as voicemail message summary information, may include information such as the length of the message, the calling party's identification, the caller's phone number, and the time and date of the voicemail. VNP 604 includes additional information as shown, such as a voicemail vendor identification, a secondary telephone access number, a message identification number, and a number of voicemail messages. As in FIG. 4, mobile device 600 has a database 607 of passwords 609 associated with each mailbox ID which the device references when a given password 609 is required by the unified messaging system for authentication.

**[0040]** FIG. 7 is an example illustration of a message format 700 used to communicate a voicemail notification message having a voicemail notification payload (VNP). Message format 700 includes a byte string that specifies all relevant information about the voicemail message, including the voicemail message summary information and the voicemail message retrieval/processing information. Message format 700 may be provided in an e-mail message or, alternatively, in a short message service (SMS) message. If provided in an e-mail message, it may be located within an attachment of the e-mail message. If provided in an SMS message, it may be located in the body of the SMS message or, alternatively, in a data header of the SMS message. For the latter in particular, GSM 3.40 9.2.3.24 provides for available data header space which may used for such voicemail notification purpose.

**[0041]** The first byte in message format 700 is a voicemail notification message code 702 used to indicate that the message pertains to a voicemail notification. As shown in this example, the value "0xC1" is designated as voicemail notification message code 702 to indicate that the message is a voicemail notification message. The second byte in message format 700 specifies a total length 704 of the message, which may be a maximum of 140 bytes. After the total length information 704, what follows is a plurality of type-length encoded fields (such as a type-length encoded field 706) most pertinent to the VNP. Following type-length encoded field 706, additional type-length encoded fields 714 are preferably provided. Each type-length encoded field 706 includes a byte code field 708, a length field 710, and a voicemail-related data field 712. Each byte code field 708 includes data which describes the type of voicemail-related data which is inserted within voicemail-related data field 712. Each length field 710 describes the length (e.g. in bytes) of the voicemail-related data which is inserted within voicemail-related data field 712.

**[0042]** The following Table 1 outlines one example of information that may appear in the VNP, such as that which may be included in message format 700 of FIG. 7. Note that the information in Table 1 corresponds to that information described in the example of FIG. 4. This data is customizable with respect to the service provider.

Table 1.

| Example of Voicemail Notification Fields. | | | |
|---|---|---|---|
| TYPE | BYTE CODE | LENGTH | DESCRIPTION |
| MAILBOX_ID | 0x91 | 8 | The unique ID of the voice mailbox |
| | | | containing the voicemail message. (long hash) |
| PROTOCOL_ID | 0x94 | 8 | The ID of the protocol being used. (long hash) |
| PROTOCOL_VERSION | 0x98 | 1 | The version of the protocol being used. |
| SECURITY_LEVEL | 0x9C | 1 | 0 indicates MSISDN only; 1 indicates MSISDN and password are required. |
| ACCESS_NUMBER | 0xB8 | 16 | The voicemail access number that is to be called in order to play the message. |
| CALLER_ID | 0xBC | 16 | The caller ID (phone number) of the caller who left the voicemail message, if available. |
| TIMESTAMP | 0xD1 | 8 | The date/time at which the message was left. (64-bit long) |

Table 1. (continued)

| Example of Voicemail Notification Fields. | | | |
|---|---|---|---|
| TYPE | BYTE CODE | LENGTH | DESCRIPTION |
| MESSAGE_LENGTH | 0xD4 | 2 | The length in seconds of the voicemail message. |

**[0043]** In Table 1, the MAILBOX_ID is a unique identifier of the service provider voice mailbox that contains the voicemail message. Eight bytes are preferably allocated for this identifier, which should be a long hash value of a string describing the mailbox. The mobile device may use the mailbox ID to determine which password to use when making a voicemail access call.

**[0044]** The PROTOCOL_ID is a unique identifier for the protocol used by the service provider. Four bytes are allocated for this ID, which should be a long hash value of the string describing the protocol. The PROTOCOL_VERSION is version of the protocol to be used for the given voicemail message. The protocol ID and version together indicate the mappings between the access control commands and the corresponding DTMF tones.

**[0045]** The SECURITY_LEVEL indicates the security level required for authentication from the service provider. For example, the provider may only require the mobile subscriber ISDN (integrate services digital network) (MSISDN) or it may require the MSISDN plus password. In one embodiment, a 0 indicates MSISDN only; a 1 indicates MSISDN plus password.

**[0046]** The ACCESS_NUMBER is the phone number to be called by the device in order to play the voicemail. The CALLER_ID is the phone number of the caller who left the voicemail message. The number must be fully qualified, including country code, area/city code, and phone number. The TIMESTAMP is the time at which the voicemail message was deposited in the voice mailbox, specified as the number of milliseconds since midnight the January 1,1970 UTC. The MESSAGE_LENGTH is length in seconds of the voicemail message.

**[0047]** The VNP may also include such information as DTMF tone lengths. The MIN_DTMF_TONE_LENGTH is the minimum length in milliseconds of an individual DTMF tone for the receiving voicemail system can correctly recognize the tone. MAX_DTMF_TONE_LENGTH is the maximum length in milliseconds of an individual DTMF tone in order that the receiving voicemail system can correctly recognize the tone. The DTMF_GAP_LENGTH is the minimum length in milliseconds of the gap between individual DTMF tones. The SKIP_LENGTH is length in seconds that playback skips forward or backward when a SKIP_FORWARD or SKIP_BACKWARD command is received by the voicemail system.

**[0048]** Referring now to the following Table 2, a different example of information that may appear in the VNP, such as that which may be included in message format 700 of FIG. 7, is shown. Note that the information in Table 2 corresponds specifically to that information described in the example of FIG. 5. This data is also customizable with respect to the service provider.

Table 2.

| An alternative example of voicemail notification data that may appear in VNP. | | | |
|---|---|---|---|
| TYPE | BYTE CODE | LENGTH | DESCRIPTION |
| MAILBOX_ID | 0x91 | 8 | The unique ID of the voice mailbox containing the voicemail message. (long hash) |
| FLAGS | 0x94 | 8 | Misc. bit flags. E.G., whether a password is required for authentication. |
| DTMF_CMDS | 0x98 | Variable length | Encoding of DTMF commands semantics. |
| ACCESS_NUMBER | 0xB8 | 16 | The voicemail access number that is to be called in order to play the message. |
| CALLER_ID | 0xBC | 16 | The caller ID (phone number) of the caller who left the voicemail message, if available. |
| TIMESTAMP | 0xD1 | 8 | The date/time at which the message was left. (64-bit long) |
| MESSAGE_LENGTH | 0xD4 | 2 | The length in seconds of the voicemail message. |

**[0049]** In Table 2, the MAILBOX_ID, SECURITY_LEVEL, ACCESS_NUMBER, CALLER_ID, TIMESTAMP, and MESSAGE_LENGTH serve the same purpose as in the previous example of VNP data. However, in this embodiment,

there is no allocation for identifying the protocol or protocol version. Instead, this VNP example specifies all the voicemail commands necessary to play and manipulate the voicemail within the VNP. In this embodiment, FLAGS are bit flags specifying various configuration options. This may indicate whether a password is required for authentication.

**[0050]** Referring now to the following Table 3, even another example of information that may appear in the VNP, such as that which may be included in message format 700 of FIG. 7, is shown. Note that the information in Table 3 corresponds specifically to that information described in the example of FIG. 6. This data is also customizable with respect to the service provider.

Table 3.

| Another alternative of voicemail notification data that may appear in the VNP. | | | |
|---|---|---|---|
| TYPE | BYTE CODE | LENGTH | DESCRIPTION |
| VENDOR_ID | 0x01 | Variable, Max 15 | Identifier representing the vendor which supplied the voicemail notification. (ASCII) |
| MAILBOX_ID | 0x02 | Variable, Max 15 | A unique id of the voice mailbox containing the voicemail message. It may be a string representing the voice mailbox. (ASCII) |
| FLAGS | 0x03 | Variable, Max 4 | Misc. bit flags. E.g., whether a password is required for authentication. (binary) |
| ACCESS_NUMBER | 0x04 | Variable, Max 15 | The voicemail access number that is to be called in order to play the message. (ASCII) |
| SECONDARY_ACCESS_NUMBER | 0x05 | Variable, Max 8 | The number which may need to be dialed upon connection to a PBX, in order to transfer the call to the voicemail system. (ASCII) |
| MESSAGE_COUNTS | 0x06 | 5 | Number of new, urgent, fax, and total messages, and max messages in mailbox. (binary) |
| PASSWORD_LIMITS | 0x07 | 2 | Min and Max password lengths. Defaults: min 4, max 7. (binary) |
| MESSAGE_ID | 0x08 | Variable, Max 8 | Unique id of the message within the voice mailbox. BCD with 0xf filler bit if the number of digits is odd. |
| DTMF_CMDS | 0x09 | Variable length | Encoding of DTMF access control tones. Defaults defined in Appendix A. |
| CALLER_ID | 0x0A | Max 15 | The caller ID (phone number) of the caller who left the voicemail message, if available. (ASCII) |
| MESSAGE_LENGTH | 0x0B | 2 | The length in seconds of the voicemail message. |
| TIMESTAMP | 0x0C | 4 | The date/time at which the message was left. (binary, specified as the number of seconds since January 1, 1970 00:00) |

Table 3. (continued)

| Another alternative of voicemail notification data that may appear in the VNP. | | | |
|---|---|---|---|
| TYPE | BYTE CODE | LENGTH | DESCRIPTION |
| DELETE_LIST | 0x0E | Variable | The list of messages that were deleted during the last subscriber session. Lists the size of a MSG_ID followed by list of MESSAGE ID's. E.g., 00178, 00179 delete confirmations would have format 0x3, 0x00, 0x17, 0x8f, 0x00, 0x17, 0x9f |
| ACK_LIST | 0x0F | Variable | The list of messages acknowledged by the mobile during the last |
| | | | subscriber session. Lists the size of a MESSAGE ID followed by a list of message ID's. E.g. 00178, 00179 would have the format 0x03 |

**[0051]** In Table 3, VENDOR_ID is a unique string representing the vendor which provided the voicemail notification. MAILBOX_ID represents the unique ID of the voice mailbox that contains the voicemail message. FLAGS are bit flags specifying various configuration options as well as characteristics of the voicemail message or the mailbox itself (e.g. whether the message is urgent, whether the user's mailbox is full, or whether a password is required for authentication once the call into the voicemail system is connected).

**[0052]** ACCESS_NUMBER is the phone number to be called in order to connect with the voicemail system and play the voicemail message. SECONDARY_ACCESS_NUMBER, which is optional, is a number to be dialed by the mobile device upon connection to the number specified in the ACCESS_NUMBER field. Dialing the secondary access number will transfer the call to the voicemail system. To connect to a corporate voicemail system, for example, the user is typically required to call a main phone number to connect to the corporate PBX, and subsequently dial an extension which transfers the call to the voicemail system.

**[0053]** MESSAGE_COUNTS is the number of new, urgent, and fax messages in the user's mailbox, as well as the number of messages in the mailbox and the maximum number of messages in the mailbox. PASSWORD_LIMITS is the minimum and maximum length of the voice mailbox password. There should be two bytes following the length byte. The first byte represents the minimum password length and second byte represents the maximum password length.

**[0054]** MESSAGE_ID is the unique ID of the message within the voice mailbox identified by MAILBOX_ID. CALLER_ID is the phone number of the caller who left the voicemail message. The number must be fully qualified, including country code, area/city code, and phone number. MESSAGE_LENGTH is the length in seconds of the voicemail message. TIMESTAMP is the time at which the voicemail message was deposited in the voice mailbox, specified as the number of seconds since midnight the January 1, 1970 UTC.

**[0055]** DELETE_LIST is a list of message IDs that have been deleted by way of a DELETE command. This field should be in the format [TYPE][LENGTH][MSGID LENGTH][ID1][ID2]...[IDN]. ID1...ION. should be nibble-packed BCD with an "f' filler bit if the number of digits in the message IDs is odd. The byte MSGID LENGTH byte indicates the number of bytes needed to represent the message ID, not the number of digits in the message ID. ACK_LIST is a list of messages that have been acknowledged by the device via the ACK command. This field is to be encoded in the same manner as the DELETE_LIST field.

**[0056]** DTMF_CMDS is a string of bytes specifying the DTMF sequences required for each supported access control command. The DTMF commands are specified in two-byte pairs, such that the first byte indicates the access control command, and the second byte is a BCD-encoding of the two-digit DTMF sequence required to invoke the command. See Table 4 below for one example of DTMF access control commands.

Table 4.

| One example of DTMF access control command definitions. | | |
|---|---|---|
| COMMAND | BYTE CODE | DEFAULT VALUE |
| SET_MSG_ID | 0x01 | 0x11 |
| PLAY | 0x02 | 0x12 |

Table 4. (continued)

| One example of DTMF access control command definitions. | | |
|---|---|---|
| COMMAND | BYTE CODE | DEFAULT VALUE |
| PAUSE_PLAYBACK | 0x03 | 0x13 |
| RESUME_PLAYBACK | 0x04 | 0x14 |
| SKIP FWD | 0x05 | 0x15 |
| SKIP_BACK | 0x06 | 0x16 |
| DELETE_MSG | 0x20 | 0x21 |
| FORWARD MSG | 0x23 | 0x22 |
| REPLY_TO_MSG | 0x24 | 0x23 |
| ACK_MSG_IDS | 0x30 | 0x24 |
| DELETE_MSG_IDS | 0x31 | 0x25 |
| PLAY GREETING | 0x40 | 0x41 |
| DELETE_GREETING | 0x41 | 0x42 |
| RECORD_GREETING | 0x42 | 0x43 |
| SET_PASSWORD | 0x43 | 0x44 |
| SET_GREETING | 0x44 | 0x45 |
| RECORD_NAME | 0x45 | 0x46 |
| DELETE_NAME | 0x46 | 0x47 |
| PLAY_NAME | 0x47 | 0x48 |

**[0057]** FIGs. 8 and 9 are basic flowcharts describing communication and processing of the voicemail notification data described in relation to FIGs. 4-7. FIG. 8 describes communication from the system to the mobile device, and FIG. 9 describes the reception and processing of such information at the mobile device. Beginning at a start block in FIG. 8, if a voicemail message is incoming as tested at step 902, then it will be received and stored at the voicemail system in a particular user's voice mailbox (step 904). In response, a voicemail notification message will be sent to a mobile device associated with the voice mailbox (step 906). The voicemail notification message includes voicemail message summary information and voicemail message retrieval/processing information as described above.

**[0058]** Beginning at a start block in FIG. 9, if a voicemail message is stored in the voicemail system at step 1002, then a voicemail notification message will be received at the mobile device shortly thereafter (step 1004). This voicemail notification message includes voicemail message summary information and voicemail message retrieval/processing information as described above, which is stored in the mobile device. The mobile device alerts the end user through its user interface (step 1006). Using the voicemail message summary information, the notification is preferably displayed in the form of a message header line in the visual display of the mobile device (step 1008). Using the voicemail message retrieval information, the mobile devices provides transparent voicemail retrieval and processing functions at the user interface (step 1010). To do this, the mobile device is preferably provided with a graphical user interface (GUI) having visual objects associated with basic voicemail functions such as PLAY, REWIND, FAST FORWARD, SKIP BACK, SKIP FORWARD, SAVE, and DELETE. Additional voicemail notifications may be subsequently received and handled similarly. Advantageously, the end user needs not remember voicemail system specific commands for each voicemail system and therefore voicemail processing is made much easier. A common user interface may be utilized for each different voicemail system in which voicemail messages are kept.

**[0059]** FIG. 10 is a flowchart which describes an overall system method for use in receiving and manipulating voicemail messages from a mobile device. In step 1000, a call is received for a user. This call may come from any source, such as a landline telephone, a corporate phone, or a mobile phone, In step 1002, a traditional private branch exchange (PBX) system sends the call to a voicemail system. In step 1004, a voicemail message from the call is recorded and saved. In step 1006, the voicemail system sends a summary of call details to the PBX interface. As checked in step 1008 if the user does not have a mobile device, the call remains in the voicemail system until the user can retrieve the call by conventional methods at an end block 1010. As checked in step 1008, if the user does have a mobile device, then the voicemail message is placed in unified message system (UMS) storage in step 1012.

**[0060]** In step 1014, the UMS sends a voicemail notification message to the mobile device. In step 1016, upon receipt of the voicemail notification message the mobile device alerts the user. In step 1018, the user selects to "open" the message and thereby issues a command to retrieve the message. In response, in step 1020, the mobile device calls the UMS using the voicemail system access number provided in the voicemail notification message. In step 1022, once the mobile device has connected to the UMS, without further user intervention the mobile device sends the DTMF tone commands that identify the particular voicemail message and subsequently the command that the user wishes to perform on the voicemail message. The DTMF tone commands associated with a particular voicemail service provider reside in a database on the mobile device. Alternatively, the DTMF tone commands may be provided in the voicemail notification payload of the voicemail notification message. In step 1024, the UMS authenticates the request when the device sends the DTMF tones representing the user's password. This is done as a security precaution.

**[0061]** In step 1026, the UMS determines whether the voicemail message is still available. If the voicemail message is not available, then in step 1028 the UMS sends the mobile device a response that the voicemail is invalid. This may be a response that is audible to the end-user (e.g. a voice response indicating: "This message is invalid. Please try again."). If the voicemail message is available, then in step 1030 the UMS executes the command on the voicemail message. Once the voicemail command is completed, in step 1032 the UMS waits for the user to issue new commands or may prompt the user for any new commands. If there are new commands in step 1032, the UMS preferably executes them in step 1030. If there are no new commands for this voicemail in step 1032, then in step 1034 the user may select a different voicemail message. If the user chooses to select a different voicemail message in step 1034, then the mobile device preferably sends the corresponding DTMF tone commands at step 1022 for the new voicemail message that the user wishes to retrieve. If the user does not wish to select another voicemail message in step 1034, the user may terminate the connection or the UMS may timeout after a period of inactivity. In either case, the session with the UMS ends.

**[0062]** FIGs 11-16 are illustrations of a visual display of a mobile device showing examples of voicemail message related data. As will be described, the mobile device preferably provides a graphical user interface (GUI) with visual objects in the visual display for end-user processing of voicemail messages. As an alternative to using a GUI at the mobile device for processing voicemail messages, mechanical switches associated with function indicators (e.g. PLAY or "→", FAST FORWARD or "→→, ", or REWIND or "←←" , etc.) may be utilized with or without displaying visual objects in the visual display.

**[0063]** In FIG. 11, an example of a message list 1108 which is displayed on a visual display 1100 of a mobile device is shown. Message list 1108 includes a plurality of message header lines including a message header line 1106 for an e-mail or SMS message and a voicemail message header line 1104 for a voicemail message. Each message header line shows message summary information from the VNP (e.g. a message time stamp and caller identification). Visual display 1100 also displays an indication of the number of new messages received 1102 at the mobile device. Although three message header lines are shown in FIG. 11, any number of message header lines may be simultaneously displayed.

**[0064]** In FIG. 12 it is shown that voicemail message header line 1104 from FIG. 11 is selected from message list 1108. When a voicemail message header line is selected, a pull-down menu 1202 of functions from which the user may select is displayed on visual display 1100. As shown, the possible functions include OPEN, PLAY, MARK OPENED, DELETE, COMPOSE E-MAIL, COMPOSE PIN, PLACE CALL, COMPOSE SMS, SEARCH, OPTIONS, and CLOSE. When a voicemail message header line is selected from message list 1108, the default option in menu 1202 is a PLAY function 1204. When PLAY function 1204 is selected, the mobile device preferably calls the appropriate voicemail telephone access number listed in the VNP and subsequently sends the appropriate DTMF tone command(s) from the VNP to play the selected voicemail message. Alternately, the user may OPEN the voicemail notification to display further information about the voicemail.

**[0065]** In FIG. 13 it is shown that voicemail message header line 1104 from FIG. 11 is "opened" to view further voicemail summary information 1302 from the VNP, such as information 1304 indicative of a message date and time and a message length, and a caller identification 1306 (e.g. name and phone number). FIG. 14 shows the function options available to the user with the opened voicemail summary information 1302. A pull-down menu shows the choices available, the default option being a PLAY function 1402. Other function options that may be available to the user are a CALL function 1404 to call (back) the caller, or an SMS send function 1406 to send an SMS message to the caller. Alternative options that may be available to the user are to send an e-mail to the caller or to send a fax to the caller. When PLAY function 1204 is selected, the mobile device preferably calls the appropriate voicemail telephone access number listed in the VNP and subsequently sends the appropriate DTMF tone command(s) from the VNP to play the selected voicemail message.

**[0066]** With respect to FIGs. 11-14, it is noted that since the each voicemail summary data is provided in its own data field in the VNP, it may be suitably manipulated and positioned as appropriate in visual display 110 by the mobile device with or without a heading or other useful surrounding explanatory text (e.g. "Length:" and "seconds" in the "Length: 6 seconds" line). This is in contrast to a message (e.g. an SMS message) which may carry data in a generic

field for visual display.

**[0067]** FIG. 15 shows visual display 1100 where the PLAY function for the voicemail message was just selected by the user. Visual display 1100 shows a number of new messages received 1504, number information 1510, and processing status information 1518, as well as calling party identification and message length information 1516 which are in a separate viewing area 1512. Visual display 1100 also may include a control bar 1506 from which the user can manipulate or process the voicemail message. Control bar 1506 has a plurality of functions 1508 from which the user may select. These functions may be in the form of visual buttons or icons and may include, but are not limited to, such functionality as PLAY, PAUSE, REWIND, FAST FORWARD, SKIP BACK, SKIP FORWARD, and HANG-UP. As an alternative to this VCR-type GUI interface, a pull-down menu offering the same functions as visual objects may be utilized. The user may scroll through these icons or buttons and highlight the button the user wishes to select. In the example shown in FIG. 15, control bar 1506 shows that "PAUSE" has been selected. A status line 1514 reflects that PAUSE has been selected. Note that the number of new messages received 1504 has changed (i.e. from "3" to "2") to reflect that this voicemail message has been reviewed.

**[0068]** FIG. 16 shows visual display 1100 when the mobile device is playing the voicemail message. When playing the voicemail message, a connect icon 1604 preferably appears which indicates the length of time that the mobile device has been connected to the unified messaging system. In this example, the user has scrolled along control bar 1506 to select FAST FORWARD function 1602. Status line 1514 subsequently changes to reflect what functionality is now highlighted on control bar 1506.

**[0069]** It is noted that voicemail functions for any voicemail message of any voicemail system may be selected by the end user at any time, even during the processing of a different voicemail message on a different voicemail system. For example, during the playing of a first voicemail message on a first voicemail system, an end user may select to play a second voicemail message on a second voicemail system. In this case, the mobile device will disconnect from the first voicemail system, call the second voicemail system, and send one or more commands to play the selected second voicemail message.

**[0070]** The above-described embodiments of the present application are intended to be examples only. Those of skill in the art may effect alterations, modifications and variations to the particular embodiments without departing from the scope of the application. For example, instead of using a GUI at the mobile device for processing voicemail messages, mechanical switches associated with function indicators (e.g. PLAY or "→", FAST FORWARD or "→→", or REWIND or "←←" , etc.) may be utilized with or without displaying visual objects in the visual display. The invention described herein in the recited claims intend to cover and embrace all suitable changes in technology.

## Claims

1. In a mobile communication device, a method for use in processing a voicemail message comprising:

   receiving, through a wireless receiver of the mobile communication device, a voicemail notification message corresponding to a voicemail message received at a voicemail system; and
   storing, in memory of the mobile communication device, voicemail message processing information for the voicemail message from the voicemail notification message.

2. The method of claim 1, wherein the voicemail message processing information further comprises:

   a voicemail access telephone number associated with the voicemail system which manages the voicemail message.

3. The method of claim 1, wherein the voicemail message processing information further comprises:

   a voicemail message identification number associated with the voicemail message.

4. The method of claim 1, wherein the voicemail message processing information further comprises:

   a voicemail vendor identification associated with the voicemail message.

5. The method of claim 1, wherein the voicemail message processing information further comprises:

   information indicative of a plurality of commands corresponding to a plurality of voicemail processing functions of the voicemail system which manages the voicemail message.

**6.** The method of claim 1, further comprising:

detecting, at a user interface of the mobile communication device, an end-user request to play the voicemail message;
causing the following acts to be performed in response to detecting the end-user request:

initiating a connection with the voicemail system with use of a voicemail access telephone number in the voicemail message processing information; and
causing one or more commands to be sent to the voicemail system to select the voicemail message to be played at the mobile communication device.

**7.** The method of claim 1, further comprising:

detecting, from a user interface of the mobile communication device, an end-user selection of a visual object comprising a PLAY icon or menu item;
causing the following acts to be performed in response to detecting the end-user selection of the visual object comprising the PLAY icon or menu item:

initiating a connection with the voicemail system with use of a voicemail access telephone number in the voicemail message processing information; and
causing one or more commands to be sent to the voicemail system to select the voicemail message to be played at the mobile communication device.

**8.** The method of claim 1, further comprising:

after receiving the voicemail notification message:

visually displaying voicemail message header information identified from the voicemail notification message in a visual display of the mobile communication device.

**9.** The method of claim 1, further comprising:

after receiving the voicemail notification message:

visually displaying, in a visual display of the mobile communication device, a calling party identification of the voicemail message identified from a caller identification data field of the voicemail notification message.

**10.** The method of claim 1, further comprising:

after receiving the voicemail notification message:

visually displaying, in a visual display of the mobile communication device, a time stamp of the voicemail message identified from a time stamp data field of the voicemail notification message.

**11.** The method of claim 1, further comprising:

causing the voicemail message to be processed at the voicemail system from the mobile communication device with use of the voicemail message processing information.

**12.** In a mobile communication device, a method for use in processing voicemail messages comprising:

receiving, through a wireless receiver of the mobile communication device, a voicemail notification message corresponding to a voicemail message received in a voicemail system;

wherein the voicemail notification message includes caller identification data in a caller identification field and time stamp data in a time stamp field;
causing voicemail message header information for the voicemail message to be displayed in a visual display of the mobile communication device; and

wherein the voicemail message header information includes a caller identification corresponding to the caller identification data and a message date and time corresponding to the time stamp data.

**13.** The method of claim 12, wherein the caller identification data comprises a caller identification (ID) telephone number of a calling party of the voicemail message.

**14.** The method of claim 12, wherein the voicemail notification message further comprises message time duration data in a time duration field, and wherein the voicemail message header information includes a message time duration corresponding to the message time duration data.

**15.** The method of claim 12, further comprising:

providing a graphical user interface (GUI) at the mobile communication device for end-user processing of a voicemail message at the mobile communication device.

**16.** The method of claim 12, wherein the voicemail notification message comprises voicemail message retrieval information including a voicemail message identification number corresponding to the voicemail message.

**17.** The method of claim 12, wherein the voicemail notification message comprises voicemail message retrieval information including a voicemail vendor identification associated with a voicemail message.

**18.** The method of claim 12, wherein the voicemail notification message comprises voicemail message retrieval information including one or more voicemail access telephone numbers associated with a voicemail message.

**19.** The method of claim 12, wherein the voicemail notification message comprises voicemail message processing information which indicates a first command code corresponding to a voicemail message PLAY command, a second command code corresponding to a voicemail message SKIP BACK command, and a third command code corresponding to a voicemail message SKIP FORWARD command.

**20.** A method of facilitating the processing of voicemail messages by a mobile communication device, the method comprising:

providing a voicemail notification message corresponding to a voicemail message received by a voicemail system;
causing the voicemail notification message to be sent to a mobile communication device; and
the voicemail notification message including voicemail message processing information associated with the voicemail message.

**21.** The method of claim 20, wherein the voicemail message processing information are included in a plurality of type-length encoded fields of the voicemail notification message.

**22.** The method of claim 20, wherein a plurality of type codes are included in the voicemail notification to identify the voicemail message processing information.

**23.** The method of claim 20, wherein the voicemail message processing information comprises:

a voicemail access telephone number associated with a voicemail system which manages the voicemail message.

**24.** The method of claim 20, wherein the voicemail message processing information comprises:

a voicemail message identification number corresponding to the voicemail message.

**25.** The method of claim 20, wherein the voicemail message processing information comprises:

information indicative of a plurality of commands corresponding to a plurality of voicemail processing functions of the voicemail system.

**26.** The method of claim 20, wherein the voicemail message processing information comprises:

a voicemail system vendor identification associated with a voicemail system which manages the voicemail message.

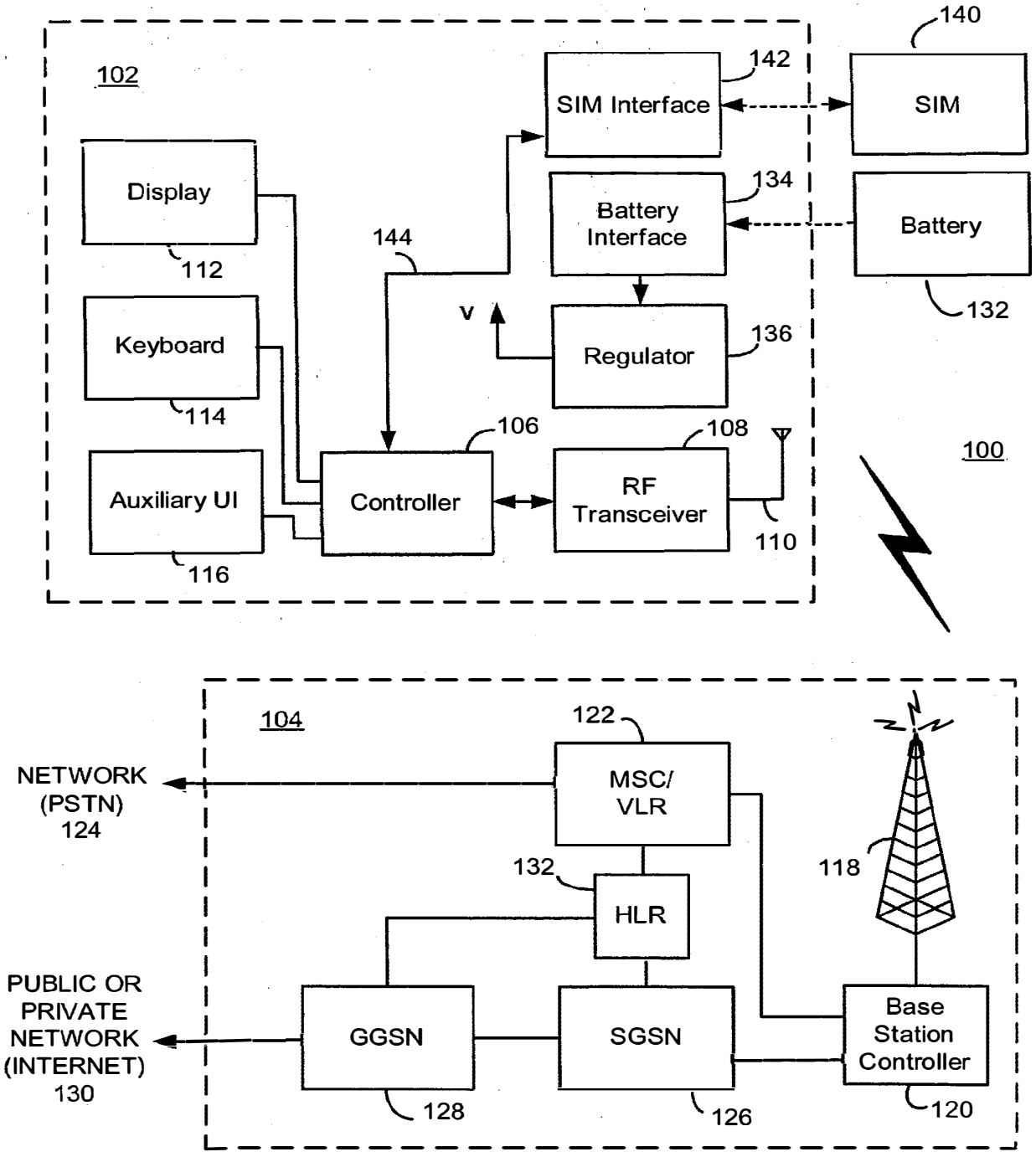
**27.** The method of claim 20, wherein the voicemail notification message further comprises voicemail summary information associated with the voicemail message comprising:

a calling party identifier associated with the voicemail message.

**28.** The method of claim 20, wherein the voicemail notification message further comprises voicemail summary information associated with the voicemail message comprising:

a time stamp of the voicemail message.

FIG. 1


102
SIM Interface
142
140
SIM
Display
112
Battery Interface
134
Battery
132
144
V
Regulator
136
Keyboard
114
106
108
100
Auxiliary UI
116
Controller
RF Transceiver
110
104
122
NETWORK (PSTN) 124
MSC/ VLR
118
132
HLR
PUBLIC OR PRIVATE NETWORK (INTERNET) 130
GGSN
SGSN
Base Station Controller
128
126
120

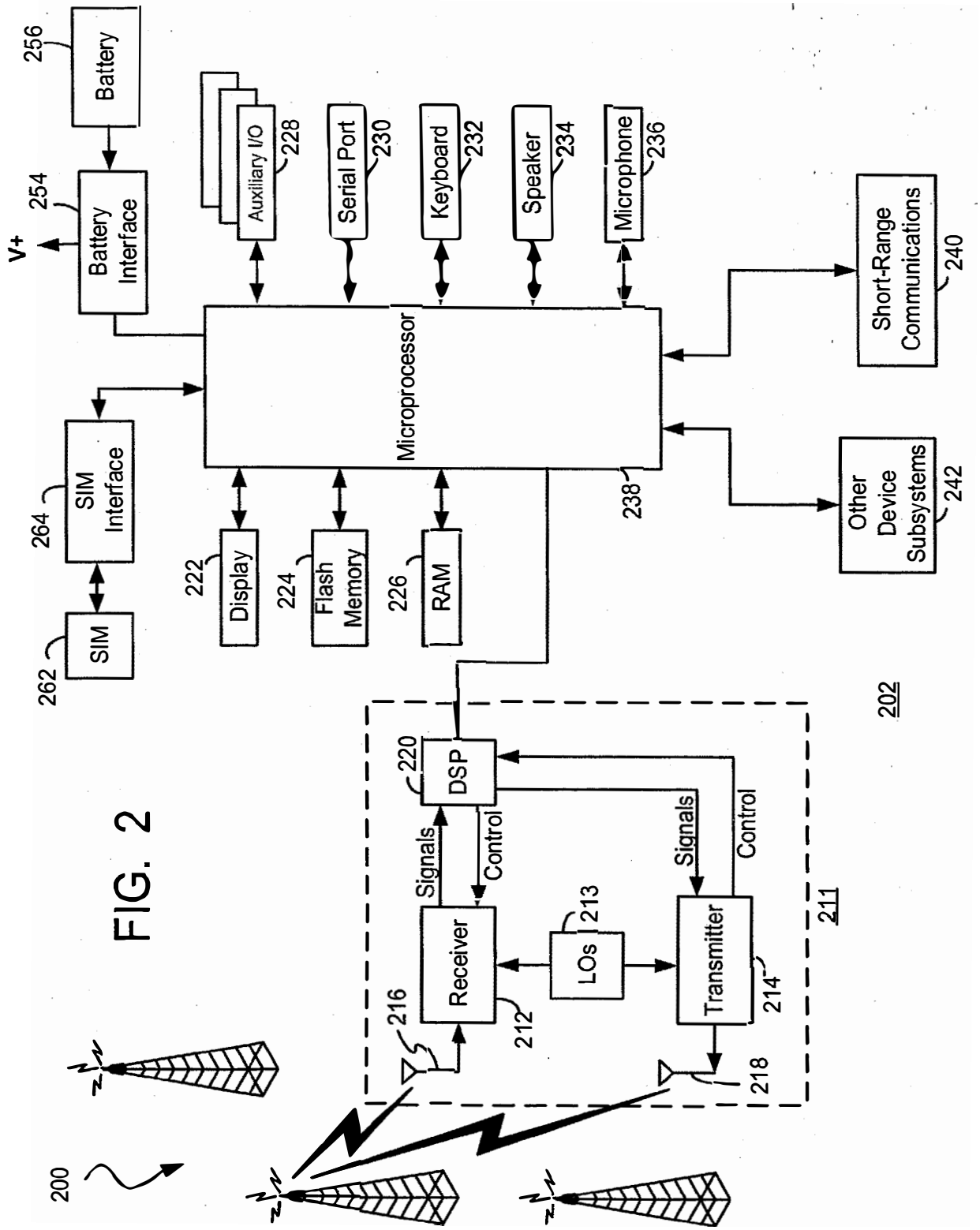

256
Battery
254
Battery Interface
V+
Auxiliary I/O
228
Serial Port
230
Keyboard
232
Speaker
234
Microphone
236
Short-Range Communications
240
Microprocessor
238
SIM Interface
264
SIM
262
Display
222
Flash Memory
224
RAM
226
Other Device Subsystems
242
202
220
DSP
Signals
Control
Signals
Control
213
LOs
Receiver
212
Transmitter
214
216
218
211
200

FIG. 2

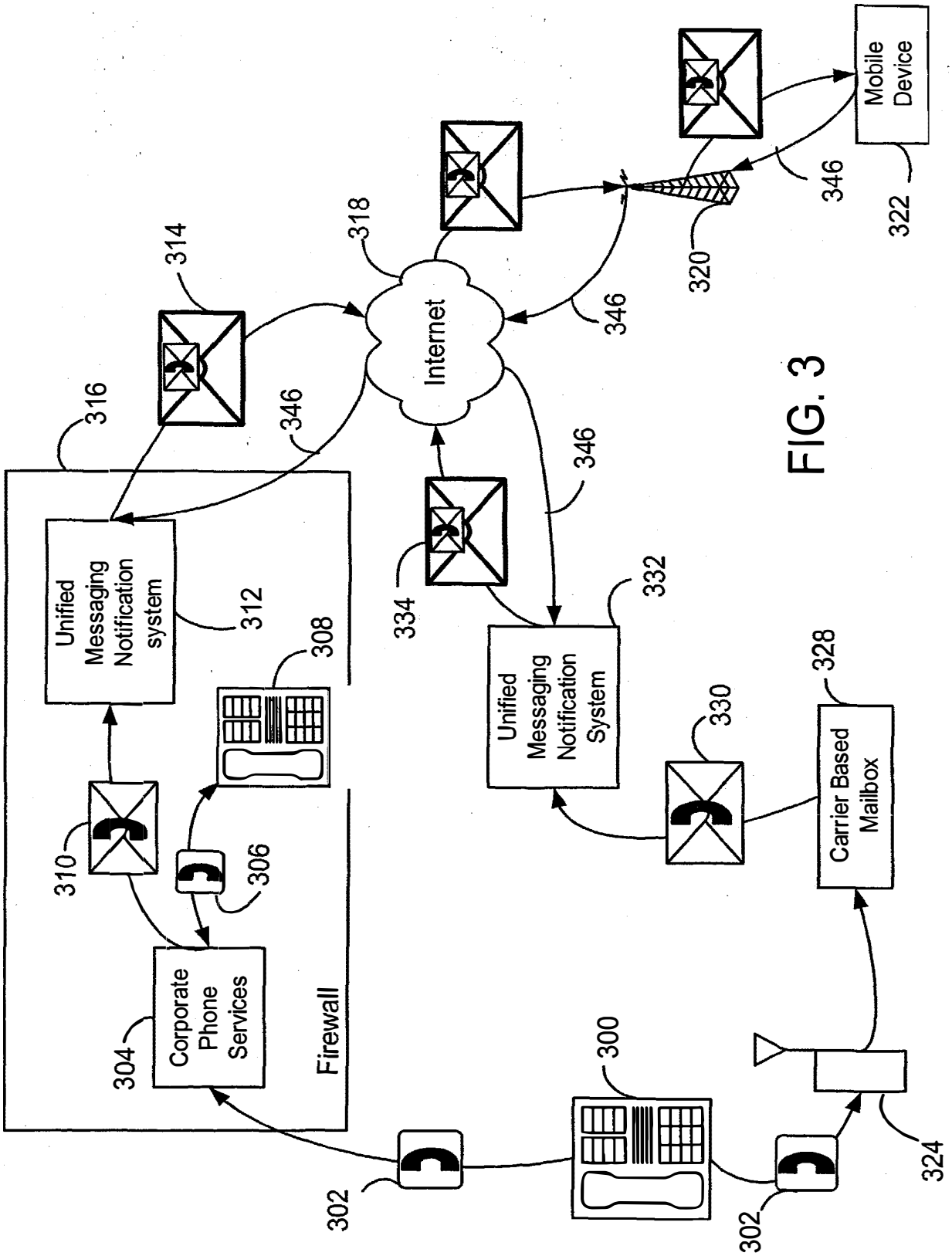

Unified Messaging Notification system
312
308
310
306
Corporate Phone Services
304
Firewall
316
314
346
Internet
318
346
346
334
Unified Messaging Notification System
332
330
328
Carrier Based Mailbox
324
300
302
302
320
346
Mobile Device
322

FIG. 3

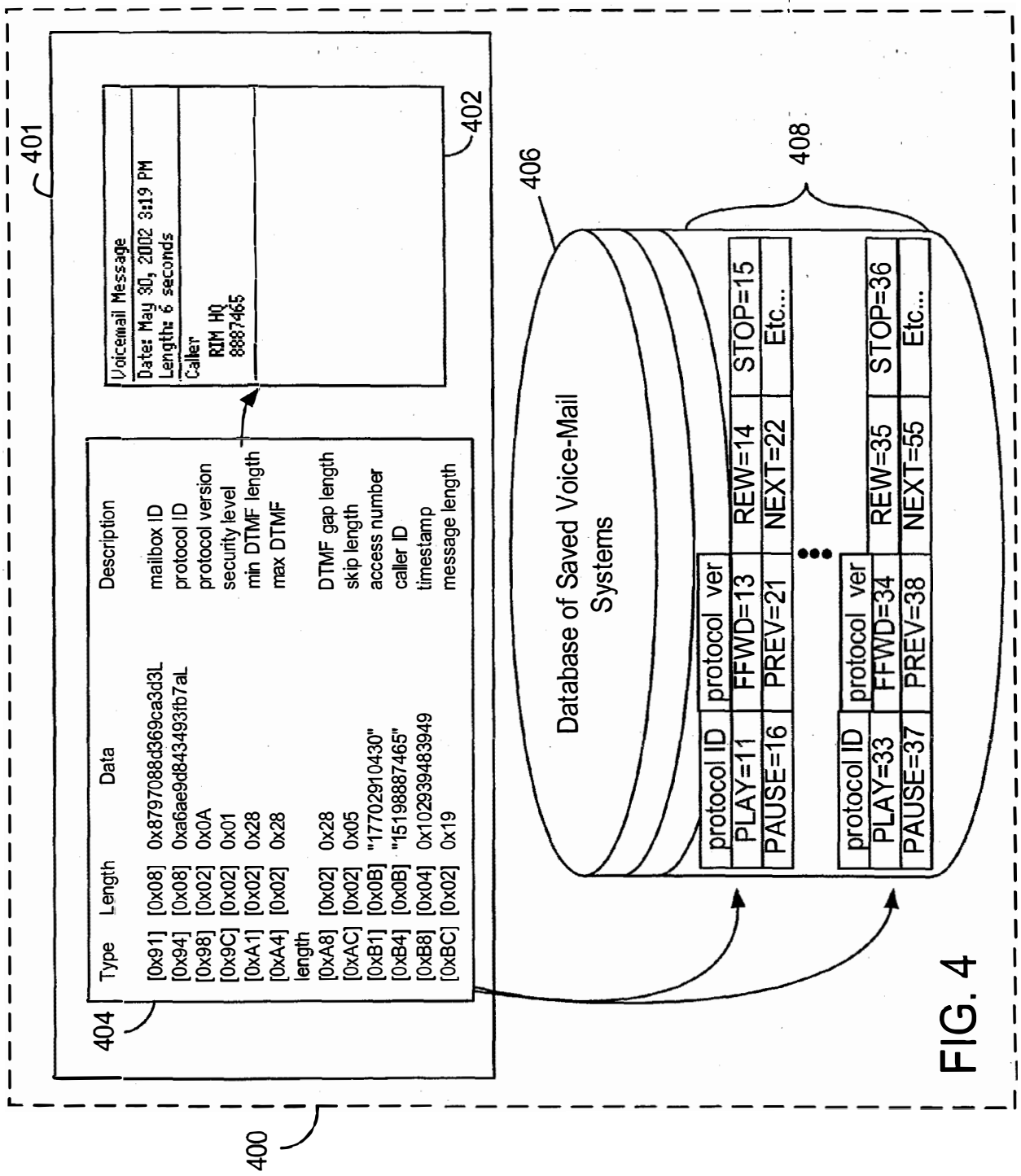

400
401
402
404
406
408
Voicemail Message
Date: May 30, 2002 3:19 PM
Length: 6 seconds
Caller
RIM HQ
88887465
Type Length Data Description
[0x91] [0x08] 0x8797088d369ca3d3L mailbox ID
[0x94] [0x08] 0xa6ae9d843493fb7aL protocol ID
[0x98] [0x02] 0x0A protocol version
[0x9C] [0x02] 0x01 security level
[0xA1] [0x02] 0x28 min DTMF length
[0xA4] [0x02] 0x28 max DTMF
length
[0xA8] [0x02] 0x28 DTMF gap length
[0xAC] [0x02] 0x05 skip length
[0xB1] [0x0B] "17702910430" access number
[0xB4] [0x0B] "15198887465" caller ID
[0xB8] [0x04] 0x102939483949 timestamp
[0xBC] [0x02] 0x19 message length
Database of Saved Voice-Mail Systems
protocol ID
protocol ver
PLAY=11
FFWD=13
REW=14
STOP=15
PAUSE=16
PREV=21
NEXT=22
Etc...
protocol ID
protocol ver
PLAY=33
FFWD=34
REW=35
STOP=36
PAUSE=37
PREV=38
NEXT=55
Etc...

FIG. 4

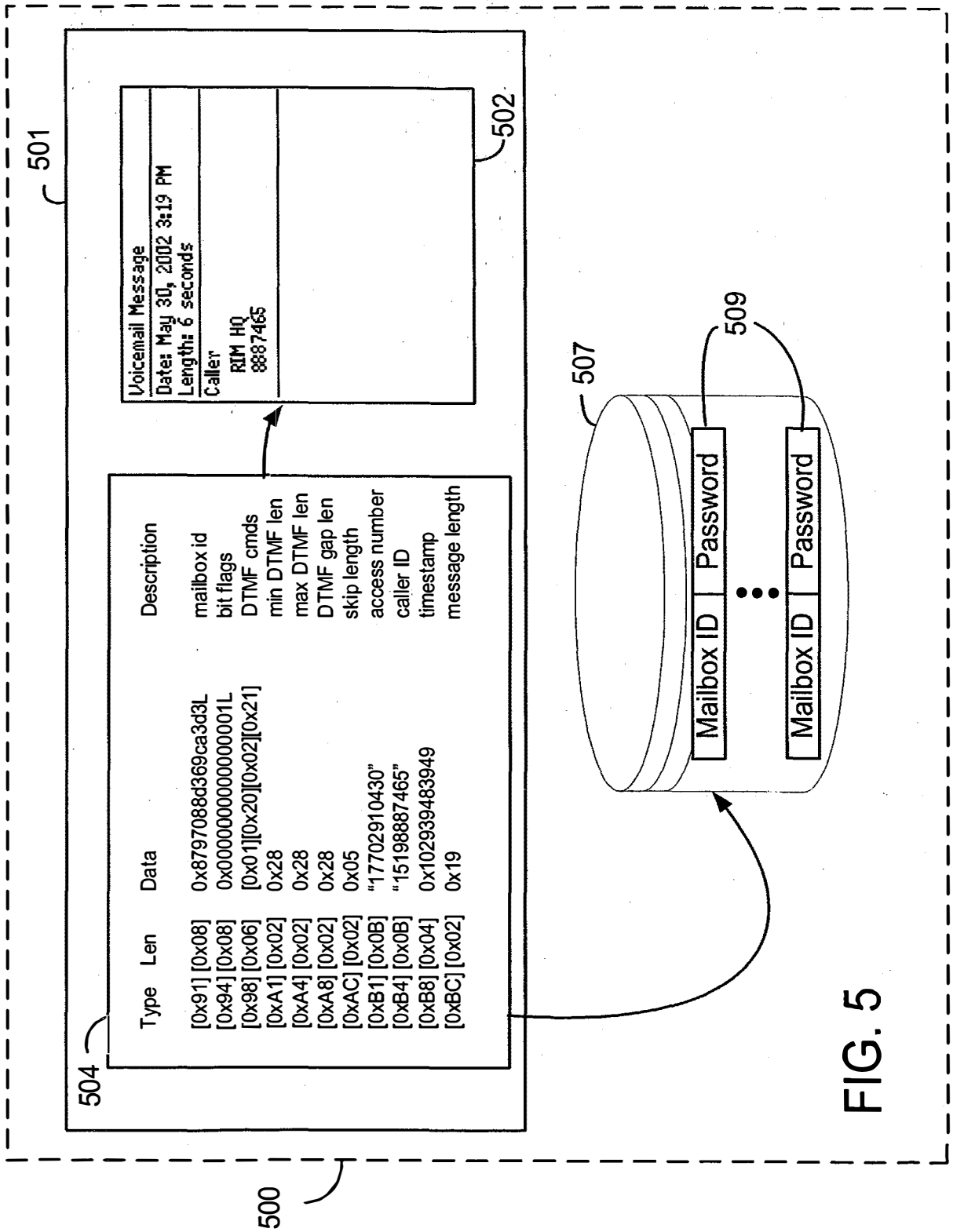

500
501
502
504
507
509
Voicemail Message
Date: May 30, 2002 3:19 PM
Length: 6 seconds
Caller
RIM HQ
8887465
Type Len Data Description
[0x91] [0x08] 0x8797088d369ca3d3L mailbox id
[0x94] [0x08] 0x0000000000000001L bit flags
[0x98] [0x06] [0x01][0x20][0x02][0x21] DTMF cmds
[0xA1] [0x02] 0x28 min DTMF len
[0xA4] [0x02] 0x28 max DTMF len
[0xA8] [0x02] 0x28 DTMF gap len
[0xAC] [0x02] 0x05 skip length
[0xB1] [0x0B] "17702910430" access number
[0xB4] [0x0B] "15198887465" caller ID
[0xB8] [0x04] 0x102939483949 timestamp
[0xBC] [0x02] 0x19 message length
Mailbox ID
Password
Mailbox ID
Password

FIG. 5

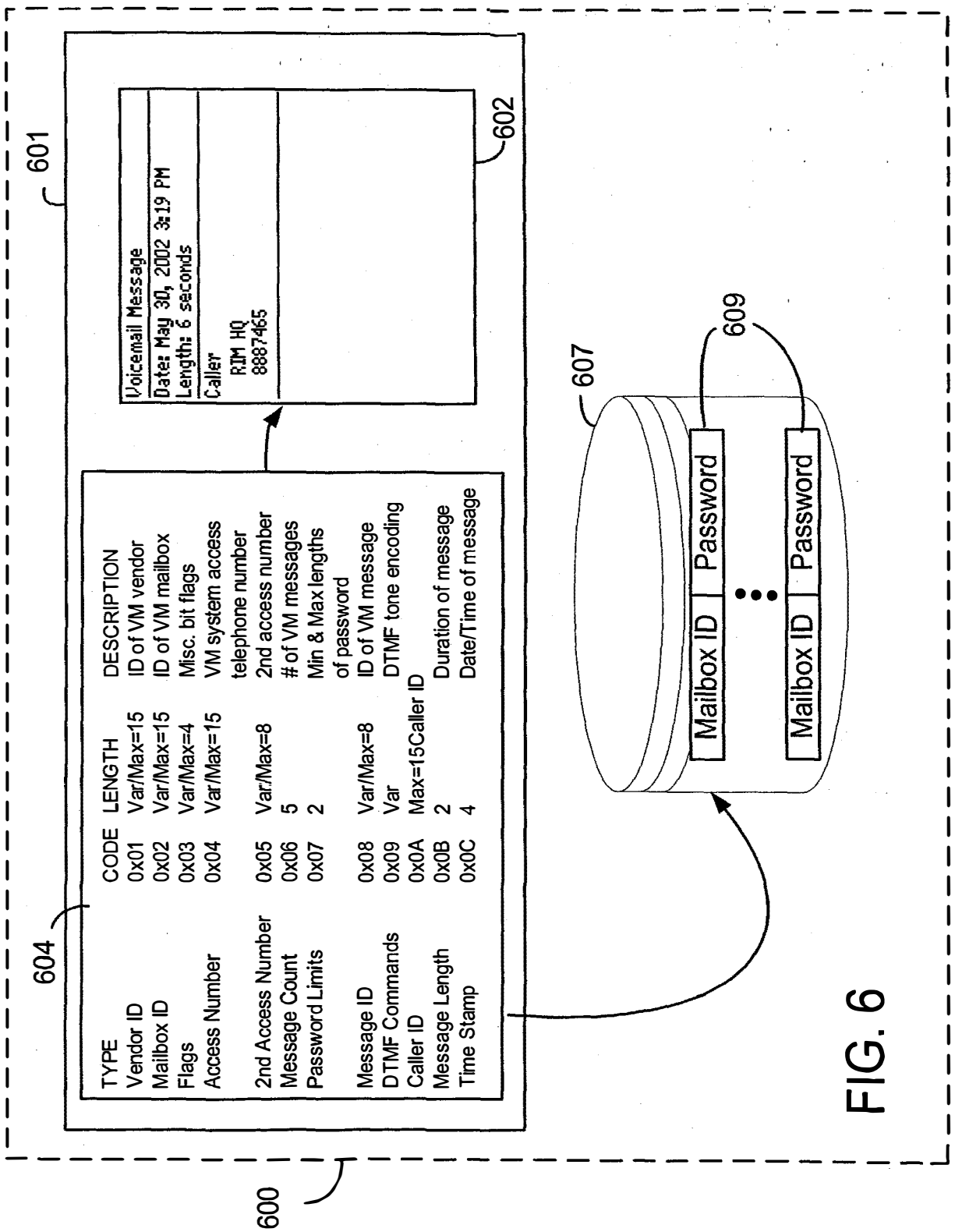

600
601
602
604
607
609
Voicemail Message
Date: May 30, 2002 3:19 PM
Length: 6 seconds
Caller
RIM HQ
8887465
TYPE CODE LENGTH DESCRIPTION
Vendor ID 0x01 Var/Max=15 ID of VM vendor
Mailbox ID 0x02 Var/Max=15 ID of VM mailbox
Flags 0x03 Var/Max=4 Misc. bit flags
Access Number 0x04 Var/Max=15 VM system access telephone number
2nd Access Number 0x05 Var/Max=8 2nd access number
Message Count 0x06 5 # of VM messages
Password Limits 0x07 2 Min & Max lengths of password
Message ID 0x08 Var/Max=8 ID of VM message
DTMF Commands 0x09 Var DTMF tone encoding
Caller ID 0x0A Max=15Caller ID
Message Length 0x0B 2 Duration of message
Time Stamp 0x0C 4 Date/Time of message
Mailbox ID
Password
Mailbox ID
Password

FIG. 6

FIG. 7

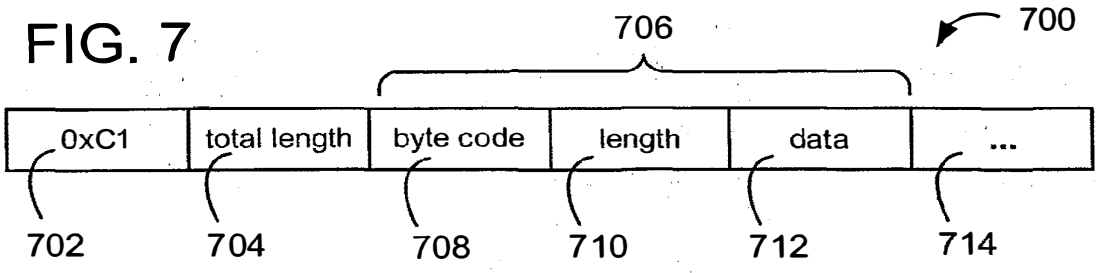



FIG. 8

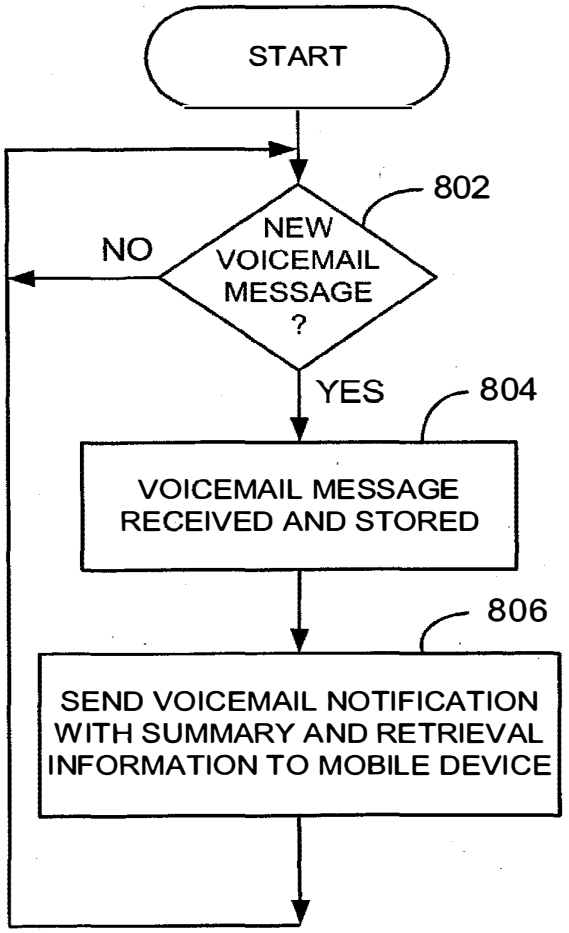



FIG. 9

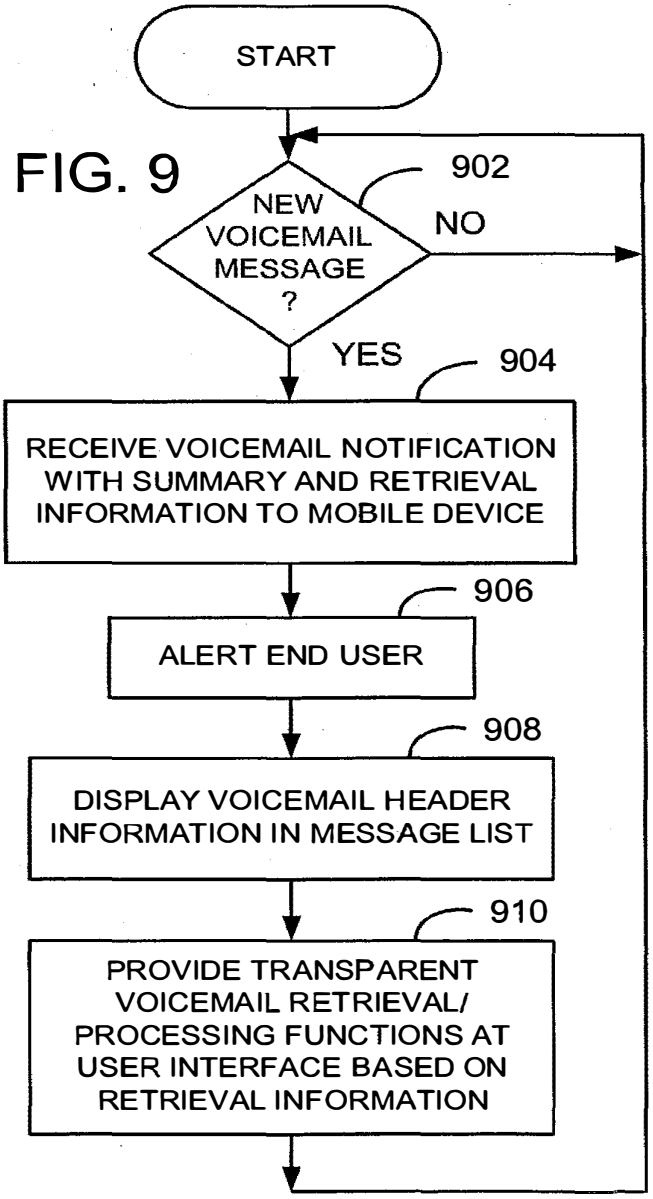


FIG. 10

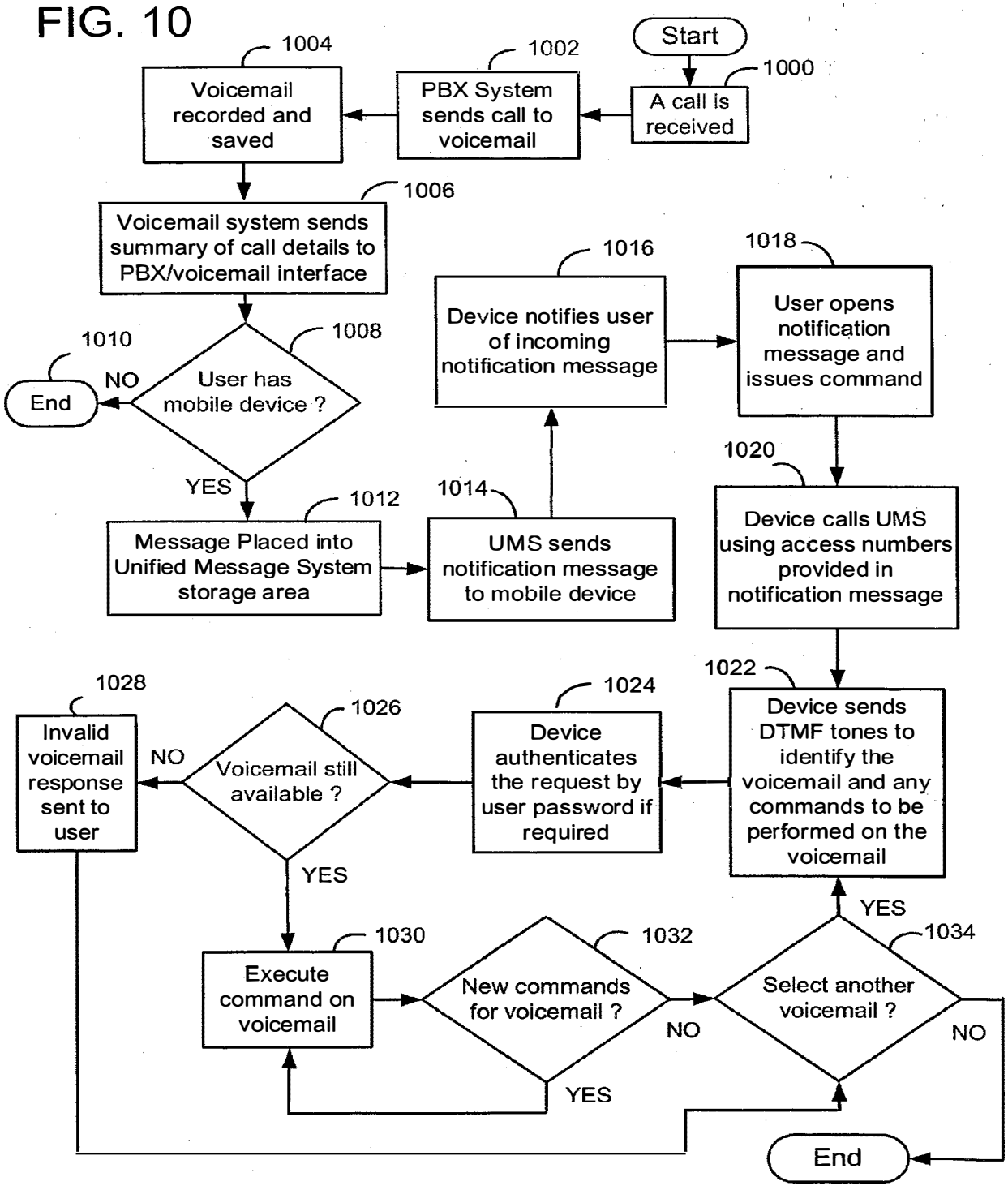

Start
1000
A call is received
1002
PBX System sends call to voicemail
1004
Voicemail recorded and saved
1006
Voicemail system sends summary of call details to PBX/voicemail interface
1008
User has mobile device ?
NO
1010
End
YES
1012
Message Placed into Unified Message System storage area
1014
UMS sends notification message to mobile device
1016
Device notifies user of incoming notification message
1018
User opens notification message and issues command
1020
Device calls UMS using access numbers provided in notification message
1022
Device sends DTMF tones to identify the voicemail and any commands to be performed on the voicemail
1024
Device authenticates the request by user password if required
1026
Voicemail still available ?
NO
1028
Invalid voicemail response sent to user
YES
1030
Execute command on voicemail
1032
New commands for voicemail ?
YES
NO
1034
Select another voicemail ?
YES
NO
End

FIG. 11

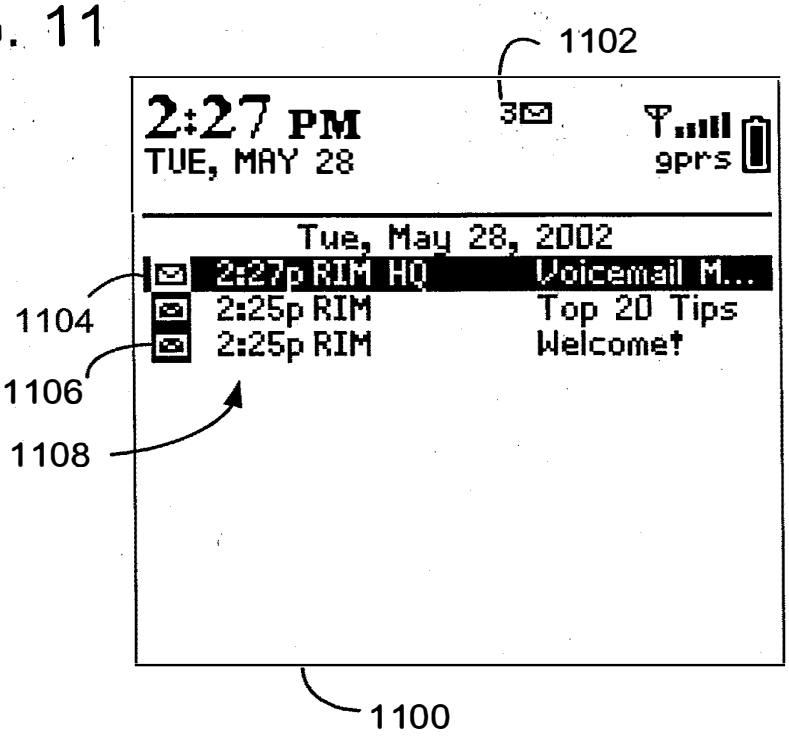

1102
2:27 PM
TUE, MAY 28
3
gprs
Tue, May 28, 2002
2:27p RIM HQ
Voicemail M...
2:25p RIM
Top 20 Tips
2:25p RIM
Welcome!
1104
1106
1108
1100

FIG. 12

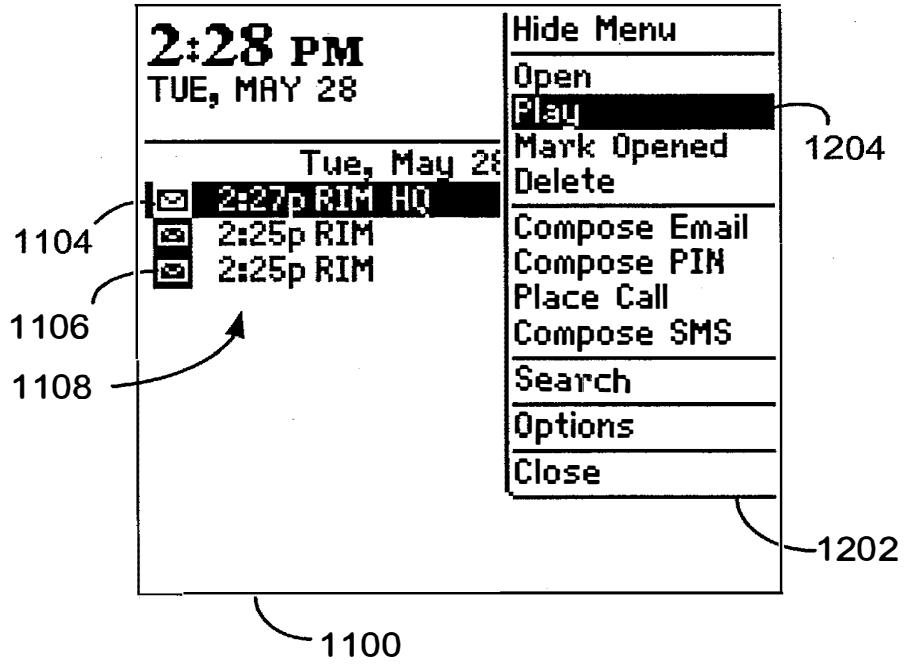

2:28 PM
TUE, MAY 28
Tue, May 28
2:27p RIM HQ
2:25p RIM
2:25p RIM
Hide Menu
Open
Play
Mark Opened
Delete
Compose Email
Compose PIN
Place Call
Compose SMS
Search
Options
Close
1204
1202
1104
1106
1108
1100

FIG. 13

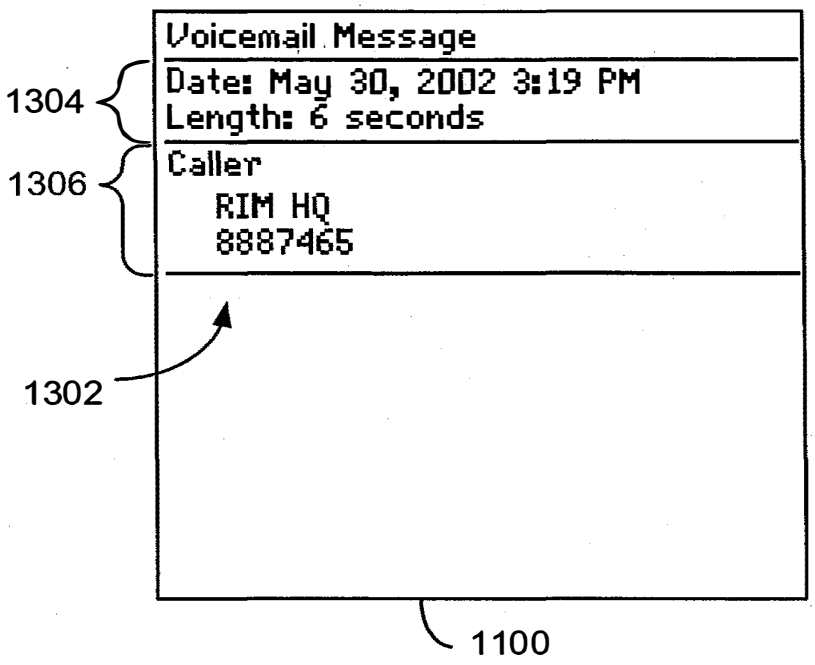

Voicemail Message
Date: May 30, 2002 3:19 PM
Length: 6 seconds
Caller
RIM HQ
8887465
1304
1306
1302
1100

FIG. 14

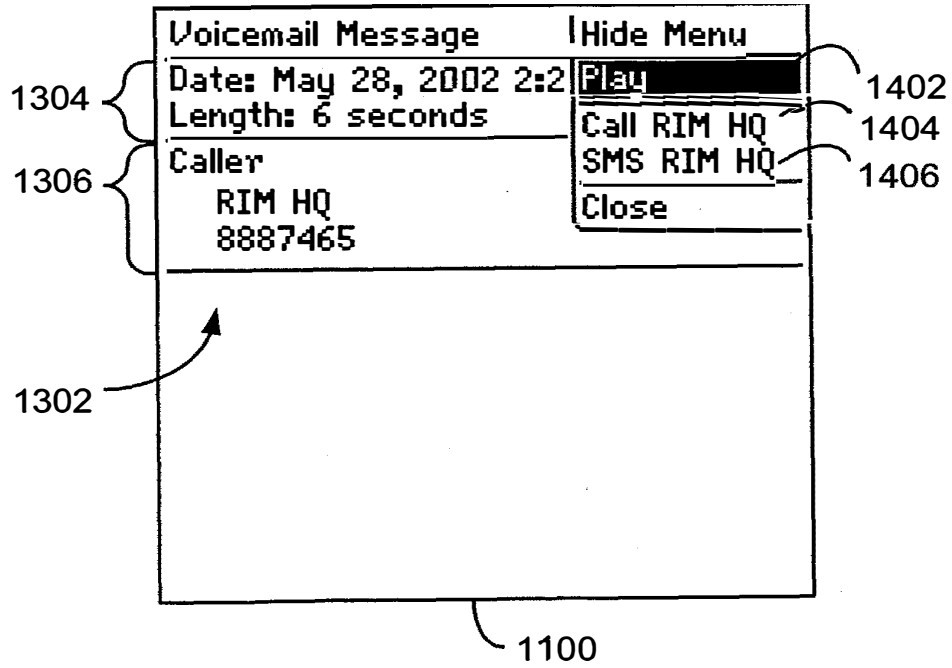

Voicemail Message
Hide Menu
Date: May 28, 2002 2:2
Length: 6 seconds
Caller
RIM HQ
8887465
Play
Call RIM HQ
SMS RIM HQ
Close
1304
1306
1302
1100
1402
1404
1406

FIG. 15

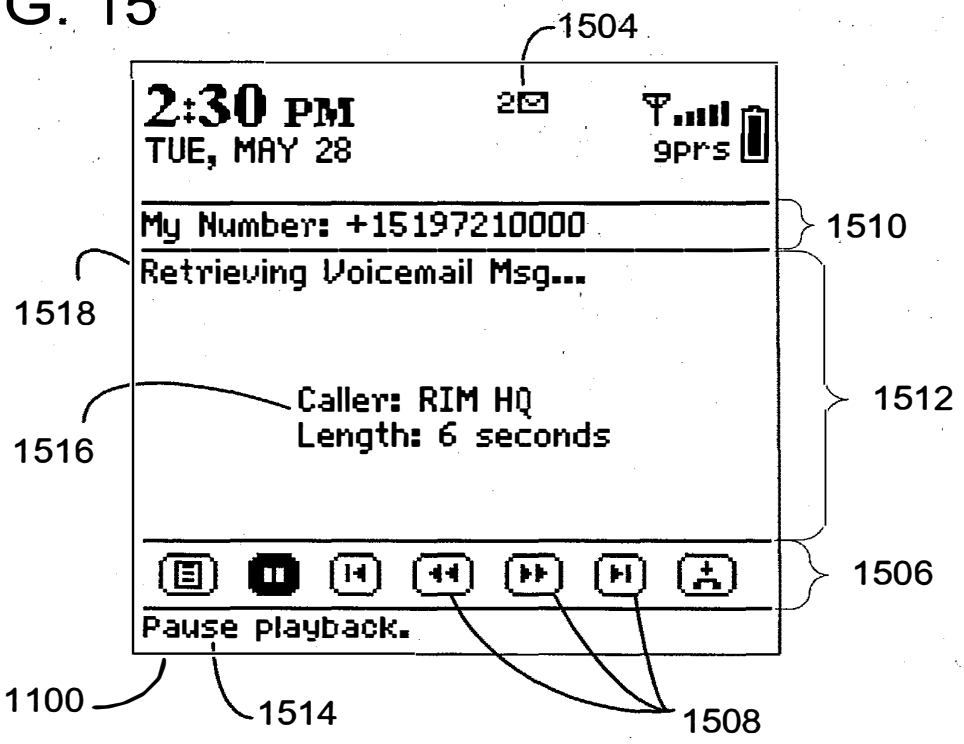
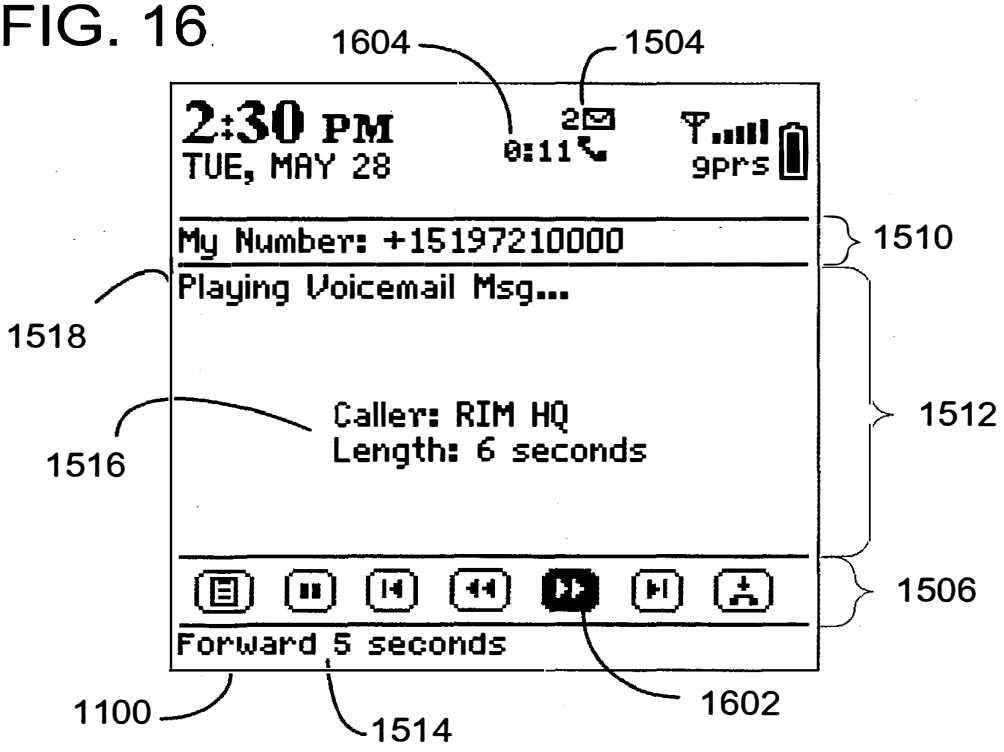

1504
2:30 PM
TUE, MAY 28
2
gprs
My Number: +15197210000
Retrieving Voicemail Msg...
Caller: RIM HQ
Length: 6 seconds
Pause playback.
1510
1512
1506
1518
1516
1100
1514
1508

FIG. 16


1604
1504
2:30 PM
TUE, MAY 28
2
0:11
gprs
My Number: +15197210000
Playing Voicemail Msg...
Caller: RIM HQ
Length: 6 seconds
Forward 5 seconds
1510
1512
1506
1518
1516
1100
1514
1602

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 715 648 A2

# (12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
**25.10.2006 Bulletin 2006/43**

(51) Int Cl.:
***H04L 29/06*** *(2006.01)*

(21) Application number: **06008224.5**

(22) Date of filing: **20.04.2006**

(84) Designated Contracting States:
**AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IS IT LI LT LU LV MC NL PL PT RO SE SI SK TR**
Designated Extension States:
**AL BA HR MK YU**

(30) Priority: **21.04.2005 KR 20050033325**

(71) Applicant: **Samsung Electronics Co., Ltd.**
**Suwon-si, Gyeonggi-Do (KR)**

(72) Inventors:
- **Lee, Seung-Yeob**
  **Yeongtong-gu,**
  **Suwon-si**
  **Gyeonggi-do (KR)**
- **Eom, Su-Hyung**
  **Yeongtong-gu,**
  **Suwon-si**
  **Gyeonggi-do (KR)**

(74) Representative: **Grünecker, Kinkeldey,**
**Stockmair & Schwanhäusser**
**Anwaltssozietät**
**Maximilianstrasse 58**
**80538 München (DE)**

## (54) Method of providing video call service in mobile station in a weak signal environment

(57) Improving the audio QoS of a video call service in a weak signal environment is accomplished. To do this, even if data is damaged in a weak signal environment during a video call using a mobile station in the weak signal environment, an audio call is preferentially processed such that a minimum audio QoS can be secured. If a transmitting mobile station (100) enters into a weak signal environment while transmitting audio data and video data (310) together during a video call, the transmitting mobile station transmits only audio data (320), and in response to this, a receiving mobile station (200) processes the audio data transmitted during a set time interval and continuously reproduces the audio data. Audio interruption can be reduced, thereby securing a minimum audio QoS.

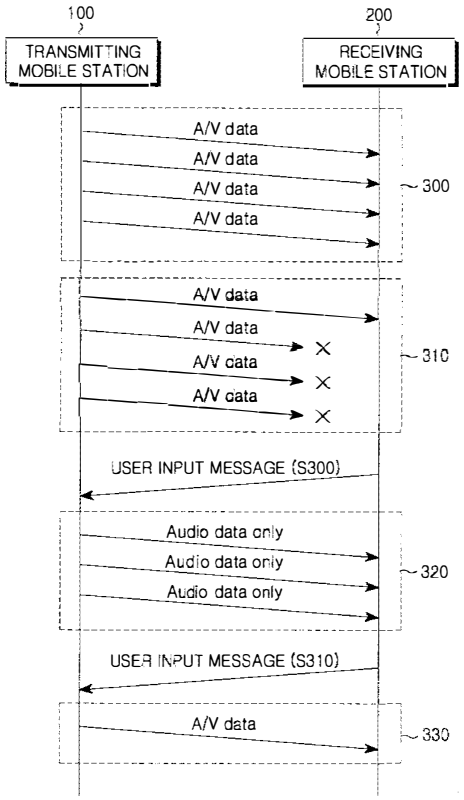



FIG.4

EP 1 715 648 A2



Printed by Jouve, 75001 PARIS (FR)

## Description

**[0001]** The present invention relates generally to a method of providing a video call service in a mobile station, and in particular, to a method of providing a video call service in a mobile station so as to preferentially guarantee an audio quality of service (QoS) in a weak signal environment during a video call.

**[0002]** Commonly, a video call service in a transmitting mobile station is achieved by compressing audio and video (A/V) data and transmitting it to a receiving mobile station. The AN data is compressed to transmit more data in a given bandwidth. In such a video call method using a mobile station, control data is inserted in the compressed AN data using a data service function and transmitted/received through a traffic channel. However, the reliability of the QoS is not guaranteed due to a delay and jitter resulting from the channel characteristics. Thus, a predetermined protocol is used in a transmitting mobile station to realize real-time streaming by detecting a data loss occurring in such a transmitting/receiving process.

**[0003]** A/V data is mainly damaged during transmission in a wireless environment, and to correct for the damage, the AN data is transmitted with control data to identify an error using a protocol such as Real-time Transport Protocol (RTP), H.324M, H.323, or Session Initiation Protocol (SIP). In particular, H.323 that is an International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) video call standard is a protocol to transmit compressed AN data in a Wideband Code Division Multiple Access (WCDMA) based transmitting mobile station. According to some of the above-described protocols, data loss can be recovered through re-transmission. However, since a video call needs real-time streaming, the protocols using the re-transmission technique are not used, and data is only minimally corrected using error correction code or discarded in a case where an error is discovered as a result of an integrity check.

**[0004]** In the WCDMA, H.324M is used to perform a video call, so the description will proceed based on H.324M, a 3rd generation (3G) video call standard protocol. A process of providing a video call service will now be described in detail with reference to FIGs. 1 and 2. FIG. 1 is a diagram for explaining a method of providing a video call service in a conventional transmitting mobile station, and FIG. 2 is a diagram for explaining a method of performing a video call in a conventional receiving mobile station.

**[0005]** Referring to FIG. 1, according to H.324M, every MUX-PDU (Multiplex-Protocol Data Unit) 20 is created by multiplexing encoded audio data 10 with video data every 20ms. Then, the transmitting mobile station transmits the MUX-PDUs 20 to an air network through traffic channels S10 to S60 at a preset time interval. In the receiving mobile station, audio data and video data are separated from AN data transmitted through the channels S10 to S60 by demultiplexing the transmitted AN data as illustrated in FIG. 2, and the separated audio data and video data are transmitted to a codec decoder outputting audio and video signals.

**[0006]** However, a video call in a handoff area or in a weak signal environment area in which the strength of an electronic wave is weak experiences occasional video freezing and audio interruption. For example, while a MUX-PDU can be normally received in a good channel state of the channel S10 and audio data A0 can be reproduced for 20ms, audio reproduction cannot be achieved for 40ms due to loss of MUX-PDUs in bad channel states of the channels S20 and S30, and then when a normal MUX-PDU is received through the channel S40, audio reproduction can be achieved for 20ms. Likewise, according channel states, audio interruption occurs by obtaining audio data A0, A3, and A5 among entire audio data A0 to A5 as illustrated in reference numeral 30 of FIG. 2.

**[0007]** From the point of view of a user, since an image output using previous data is continuously displayed even if video data is damaged, the user can continue to view the other party's face although it is a frozen image. However, audio interruption considerably degrades the service, and if the audio interruption exceeds a predetermined level, the audio is hardly audible. This phenomenon makes QoS of video data seem better than QoS of audio data from the point of view of the user. However, from the point of view of a mobile station transmitting/receiving data via the air, since audio and video data is combined single data, the QoS of the audio and video data are the same. That is, damage to AN data means that both the audio data and the video data are damaged, and since there is no method of performing quality management by separating the AN data, reliability of QoS of both the audio data and the video data is degraded.

**[0008]** As described above, in a conventional video call, the reliability of the QoS according to channel states is considerably degraded, and therefore not only the video transmission but also an audio call may not be performed. Accordingly, the QoS of a video call and call satisfaction of a user are reduced compared to a conventional audio call.

**[0009]** The present invention is for substantially solving at least the above problems and/or disadvantages and for providing at least the advantages below.

**[0010]** Accordingly, the object of the present invention is to provide a mobile station and a method of providing a video call service in a mobile station to guarantee an audio quality of service (QoS) in a weak signal environment during a video call.

**[0011]** This object is solved by the subject matter of the independent claims.

**[0012]** Preferred embodiments are defined in the dependent claims.

**[0013]** According to one aspect of the present invention, there is provided a method of providing a video call service in a mobile station in a weak signal environment, the method includes determining, by a receiving mobile

station, a state of a weak electric field based on whether audio and video (AN) data transmitted from a transmitting mobile station is damaged during a video call; if there is a weak signal environment, requesting, by the receiving mobile station, that audio data is preferentially transmitted; generating, by the transmitting mobile station, a transmittable amount of audio data in response to the request and repeatedly transmitting the audio data a predetermined number of times; and when the audio data is received, processing, by the receiving mobile station, the audio data so that the audio data is continuously reproduced.

**[0014]** The present invention will become more apparent from the following detailed description when taken in conjunction with the accompanying drawings in which:

FIG. 1 is a diagram for explaining a method of providing a video call service in a conventional transmitting mobile station;

FIG. 2 is a diagram for explaining a method of performing a video call in a conventional receiving mobile station;

FIGs. 3A and 3B are block diagrams of a transmitting mobile station and a receiving mobile station according to a preferred embodiment of the present invention;

FIG. 4 is a signaling diagram for explaining a process of providing a video call service in a weak signal environment according to a preferred embodiment of the present invention;

FIG. 5 is a diagram for explaining a method of performing a video call in an audio only mode in the transmitting mobile station according to a preferred embodiment of the present invention; and

FIG. 6 is a diagram for explaining a method of performing a video call in the audio only mode in the receiving mobile station according to a preferred embodiment of the present invention.

**[0015]** Preferred embodiments of the present invention will be described herein below with reference to the accompanying drawings. In the drawings, the same or similar elements are denoted by the same reference numerals even though they are depicted in different drawings. In the following description, well-known functions or constructions are not described in detail since they would obscure the invention in unnecessary detail.

**[0016]** In the present invention, a function of improving the audio QoS of a video call service in a weak signal environment is implemented. To do this, in the present invention, even if data is inevitably damaged in a weak signal environment during a video call using a mobile station in the weak signal environment, an audio call is preferentially processed such that a minimum audio QoS can be secured. Thus, if a transmitting mobile station enters into a weak signal environment while transmitting audio data and video data together during a video call, the transmitting mobile station transmits only audio data, and in response to this, a receiving mobile station processes audio data transmitted during a preset time interval and continuously reproduces the audio data. By doing this, audio interruption can be reduced, thereby securing a minimum audio QoS.

**[0017]** The components and operation of a mobile station in which the above-described function is implemented will now be described with reference to FIG. 3A and FIG. 3B. FIGs. 3A and 3B are block diagrams of a transmitting mobile station and a receiving mobile station according to a preferred embodiment of the present invention. Hereinafter, mobile stations having a video call function WCDMA type mobile stations based on H.324M.

**[0018]** FIG. 3A is illustrates the configuration of a transmitting mobile station and a receiving mobile station required in an excellent communication environment, FIG. 3B illustrates the configuration of a transmitting mobile station and a receiving mobile station according to a preferred embodiment of the present invention in a bad communication environment.

**[0019]** In an excellent communication environment, when a user of a transmitting mobile station 100 makes a video call, an audio call path and a video data call path are established, and audio and video are transmitted to a receiving mobile station 200 through the microphone and camera equipped in the transmitting mobile station 100. Herein, in the transmitting mobile station 100, audio data input through the microphone is encoded by an audio codec 130, and video data input through the camera is encoded by a video codec 140. Then an H.223 based AL (Adaptation Layer) 120 receives the encoded data and adds to the encoded data a sequential number indicating the transmission order.

**[0020]** A MUX layer 110 generates a series of data stream by multiplexing the encoded data, adds Cyclic Redundancy Checking (CRC) to the encoded data so that the receiving mobile station 200 can determine if an error exists in the data transmitted form the transmitting mobile station 100 through a channel in order to allow the receiving mobile station 200 to demultiplex the data stream, and then transmits the CRC added data stream to the receiving mobile station 200 through the air, i.e., a wireless network. Since this process is described in an H.223 specification, which is one of components of the H.324M protocol, its detailed description is omitted. As described above, in an excellent communication environment, the transmitting mobile station 100 is configured to include the video codec 140, the audio codec 130, the AL 120, and the MUX layer 110 illustrated in FIG. 3A.

**[0021]** Similar to the transmitting mobile station 100 in an excellent communication environment, the receiving mobile station 200 is configured to include a MUX layer 210, an AL 220, an audio codec 230, and a video codec 240 illustrated in FIG. 3A. Thus, when single data in which AN data is multiplexed is transmitted from the transmitting mobile station 100, the MUX layer 210 of the receiving mobile station 200 demultiplexes the transmitted data using the header of the transmitted data and determines

if the transmitted data is successfully received by determining if an error exists in the transmitted data using a sequential number and CRC. If the transmitted data is successfully received, the transmitted data is divided into audio data and video data, transmitted to the audio codec 230 and the video codec 240 through the AL 220, respectively, and decoded. When the audio data and the video data are synchronized and output through the above process, a video call between the transmitting mobile station 100 and the receiving mobile station 200 is achieved.

**[0022]** A detailed example to which a preferred embodiment of the present invention is applied will now be described with reference to FIG. 4. FIG. 4 is a signaling diagram for explaining a process of providing a video call service in a weak signal environment according to a preferred embodiment of the present invention.

**[0023]** In the present embodiment, if a signal is weak or in an unstable state during a video call using a WCDMA mobile station with a limited band with, e.g., 64kbps, the mobile station stops the transmission of the AN data and instead allocates a band for audio data transmission wider such that audio data can be preferentially transmitted. By doing this, by repeatedly transmitting audio data through a wider band, a data loss can be reduced in a receiving mobile station, thereby securing audio QoS.

**[0024]** Referring to FIG. 4, in step 300, the transmitting mobile station 100 transmits to the receiving mobile station 200 single data in which audio data and video data is mixed during a first time interval. If a network state is stable, e.g., if a channel state is excellent, the receiving mobile station 200 normally receives the AN data without data loss and performs a seamless video call including audio and video. However, even if the transmitting mobile station 100 transmits AN data in the same way, if the network state is unstable, in step 310, the amount of data normally transmitted to the receiving mobile station 200 is reduced. Accordingly, video freezing and audio interruption occur.

**[0025]** To prevent this phenomenon, the receiving mobile station 200 monitors data transmitted from the transmitting mobile station 100, and if the amount of received data is below a threshold, or if the number of error determinations using control data included in the received data exceeds a threshold, the receiving mobile station 200 determines that it is located in a weak signal environment in which the strength of an electronic signal is weak. Herein, since a data rate of data received by the receiving mobile station 200 is almost fixed as 64kbps, it is possible to monitor the data, which is transmitted during a set time interval, every set time interval. Thus, the receiving mobile station 200 can easily determine whether a wireless environment is deteriorated based on the monitored channel state. If the receiving mobile station 200 recognizes that it is in a weak signal environment, in step S300, the receiving mobile station 200 transmits a user input message to the transmitting mobile station 100 to request that audio data is preferentially transmitted. Herein, the structure of the user input message refers to an H.245 specification.

**[0026]** When the receiving mobile station 200 informs the transmitting mobile station 100 of a state of a weak electric field, in step 320, the transmitting mobile station 100 stops the transmission of the video data by changing a current mode in which AN data is transmitted together to an audio only mode and transmits only the audio data. As soon as changing to the audio only mode, the transmitting mobile station 100 accumulates in a local buffer the audio data to be transmitted for as many repeated transmissions required and then transmits the audio data.

**[0027]** The above-described process will now be described in more detail with reference to FIG. 5. FIG. 5 is a diagram for explaining a method of performing a video call in the audio only mode in the transmitting mobile station 100 according to a preferred embodiment of the present invention. Referring to FIG. 5 and FIG. 3B, when changing to the audio only mode, the transmitting mobile station 100 encodes audio data 500 to be transmitted using the audio codec 130. The encoded audio data 500 is output to the MUX layer 110 through the AL 120. The MUX layer 110 generates a predetermined number of MUX-PDUs by combining each of audio data so that the amount of data to be transmitted does not exceed 160 bytes. For example, when the transmitting mobile station 100 intends to repeatedly transmit data 5 times a weak signal environment, transmitting mobile station 100 can combine and transmit 5 frames of audio data. Accordingly, the MUX layer 110 can generate a single MUX-PDU with audio data A0, A1, A2, A3, and A4 of a 20ms unit as a reference numeral 510.

**[0028]** In particular, the audio only mode is performed only in a weak signal environment, and in a weak signal environment, if only a single MUX-PDU is transmitted, the receiving mobile station 200 may not receive the MUX-PDU. Thus, the transmitting mobile station 100 repeatedly transmits the same MUX-PDU during a set time interval. When 5 MUX-PDUs including the same audio data A0, A1, A2, A3, and A4 are generated as a reference numeral 520, the transmitting mobile station 100 continuously transmits each MUX-PDU 5 times within a 20ms interval. When a single MUX-PDU is generated by combining 5 frames of the audio data, the audio data is continuously accumulated in the local buffer. Thus, after the MUX layer 110 generates the MUX-PDU by combining the 5 frames of the audio data and repeatedly transmits the MUX-PDU 5 times, the MUX layer 110 repeatedly generates a MUX-PDU 530 with audio data A5, A6, A7, A8, and A9 corresponding to next 5 frames. The TX (Transmit) MUX-PDU 540 composed of MUX-PDUs generated through the above-described process is transmitted through relevant traffic channel S500 to S550 during a set time interval.

**[0029]** FIG. 6 is a diagram for explaining a method of performing a video call in the audio only mode in the receiving mobile station 200 according to a preferred em-

bodiment of the present invention. Referring to FIG. 6, when the receiving mobile station 200 receives data from the transmitting mobile station 100, the receiving mobile station 200 checks the integrity of the data, i.e., determines if an error exists in the data using CRC obtained by the MUX layer 210 and the AL 220 of FIG. 3B. If abnormal data is transmitted, the receiving mobile station 200 discards the abnormal data and decodes only normal data using the audio codec 230. Herein, since audio data is repeatedly transmitted, the MUX layer 210 stores normally transmitted data, and only if subsequent data is not identical to previous data, the MUX layer 210 transmits the subsequent data to the audio codec 230 through the AL 220.

**[0030]** In detail, the receiving mobile station 200 receives MUX-PDUs through the traffic channels S500 to S550 during a set time interval, e.g., every 20ms, as illustrated in FIG. 6. The MUX-PDUs transmitted through the traffic channels S500 to S550 are a MUX-PDU1 to a MUX-PDU6 in FIG. 6. Herein, even if the transmitting mobile station 100 sequentially transmits data through the traffic channels S500 to S550, since a wireless environment is deteriorated, the receiving mobile station 200 may receive some of the data. That is, in FIG. 6, data through the traffic channels S510, S520, and S540 is not normally transmitted, or an error is generated in each data through the traffic channels S510, S520, and S540. On the contrary, data is normally transmitted through the traffic channels S500, S530, and S550.

**[0031]** When it is determined that the MUX-PDU1 transmitted through the traffic channel S500 is normal data using CRC, decoding of the MUX-PDU1 is performed, and the audio data A0, A1, A2, A3, and A4 of a 20ms unit are output. By doing this, the audio of a total of 100ms is reproduced. The receiving mobile station 200 stores the reproduced audio data, and when it is determined that the MUX-PDU4 transmitted through the traffic channel S530 is normal data using CRC, the receiving mobile station 200 compares MUX-PDU1 with MUX-PDU4. That is, the receiving mobile station 200 compares whether the audio data of the currently received MUX-PDU4 is identical to audio data of the previously received MUX-PDU 1. If they include the same audio data, the receiving mobile station 200 ignores the currently received MUX-PDU4. When MUX-PDU6 is received through the traffic channel S550, the receiving mobile station 200 compares MUX-PDU1 with MUX-PDU6. If the audio data of MUX-PDU6 is different from the audio data of MUX-PDU1, i.e., if they are not duplicated, the receiving mobile station 200 recognizes that the new audio data is received and reproduces the new audio data A5, A6, A7, A8, and A9. By doing this, the receiving mobile station 200 can output seamless audio data without audio interruption. As a result, by repeatedly receiving a MUX-PDU from the transmitting mobile station 100 for a predetermined number of times, the receiving mobile station 200 can prevent an audio data loss the most and continuously reproduce by arranging the audio data.

**[0032]** While the audio only mode is being performed, if the strength of an electronic signal returns to the normal state due to movement of the user or other reasons, the wireless environment can become excellent. When the amount of data transmitted from the transmitting mobile station 100 returns to the normal state, or when the number of errors is reduced, the receiving mobile station 200 can recognize that it is not longer in a weak signal environment. In step S310 of FIG. 4, the receiving mobile station 200 transmits a user input message to the transmitting mobile station 100 to request that the audio only mode is released since the network state is stable. Of course, S310 can be done automatically by the receiving mobile station 200. When the transmitting mobile station 100 receives the audio only mode release request message, the transmitting mobile station 100 returns to a previous state and performs the mode in which A/V data is transmitted together. In step 330 of FIG. 4, the transmitting mobile station 100 transmits AN data together.

**[0033]** As described above, in the inventive method, by preferentially and repeatedly transmitting audio data when in a weak signal environment occurs, audio interruption can be reduced, thereby securing a minimum audio QoS.

**[0034]** While the invention has been shown and described with reference to a certain preferred embodiment thereof, it will be understood by those skilled in the art that various changes in form and details may be made therein without departing from the scope of the invention as defined by the appended claims.

## Claims

1. A method of providing a video call service in a mobile station in a weak signal environment, the method comprising the steps of:

   determining, by a receiving mobile station, whether the receiving mobile station is in a weak signal environment based on the quality of audio and video data transmitted from a transmitting mobile station during a video call;
   if it is determined that the mobile station is in a weak signal environment, requesting, by the receiving mobile station, that audio data is preferentially transmitted;
   generating, by the transmitting mobile station, a transmittable amount of audio data in response to the request and repeatedly transmitting the audio data for a preset time; and
   when the audio data is received, processing the audio data by the receiving mobile station so that the audio data is continuously reproduced.

2. The method of claim 1, wherein the step of determining if a mobile station is in a weak signal environment comprises the steps of:

determining if an error exists in data repeatedly transmitted from the transmitting mobile station; and
recognizing that the receiving mobile station is in the state of a weak electric field if the number of errors exceeds a threshold.

**3.** The method of claim 1, wherein the step of determining if a mobile station is in a weak signal environment comprises the steps of:

determining the amount of data repeatedly transmitted from the transmitting mobile station; and
recognizing that the receiving mobile station is in the state of a weak electric field if the amount of received data is below a threshold.

**4.** The method of claim 1, further comprising the steps of:

determining the amount of audio data transmitted from the transmitting mobile station;
transmitting to the transmitting mobile station a message notifying that the receiving mobile station no longer in a weak signal environment if the amount of received audio data exceeds a threshold; and
returning to a mode in which the audio and video data is transmitted together.

**5.** The method of one of claims 1 to 4, wherein in the step of requesting that audio data is preferentially transmitted, a user input message for informing of the state of a weak electric field is transmitted.

**6.** The method of claim 5, wherein the user input message is based on an H.245 specification.

**7.** The method of claim 5 or 6, further comprising the steps of:

changing to an audio only mode when the user input message is received;
encoding audio data;
assigning a sequential number to the encoded audio data, performing cyclic redundancy checking for error checking, and adding a generated cyclic redundancy checking result to the encoded data; and
accumulating the audio data, multiplexing the accumulated audio data, and transmitting the same multiplexed audio data for a predetermined number of times during a set time interval.

**8.** The method of claim 7, wherein the accumulated audio data does not exceed 160 bytes according to an H.324M specification.

**9.** The method of claim 1, wherein the step of processing the audio data so that the audio data is continuously reproduced comprises the steps of:

checking for errors in the audio data received from the transmitting mobile station;
reproducing normal audio data;
comparing the reproduced previous audio data with subsequent audio data received during the set time interval; and
reproducing the received audio data so as to follow the previous audio data if the received audio data is different from the previous audio data.

**10.** The method of claim 9, wherein the step of checking for errors is achieved by performing cyclic redundancy checking on the received audio data.

**11.** The method of claim 9, further comprising the step of:

discarding the received audio data if the received audio data is essentially equal to the previous audio data.

**12.** A mobile station adapted to operate according to one of claims 1 to 11.

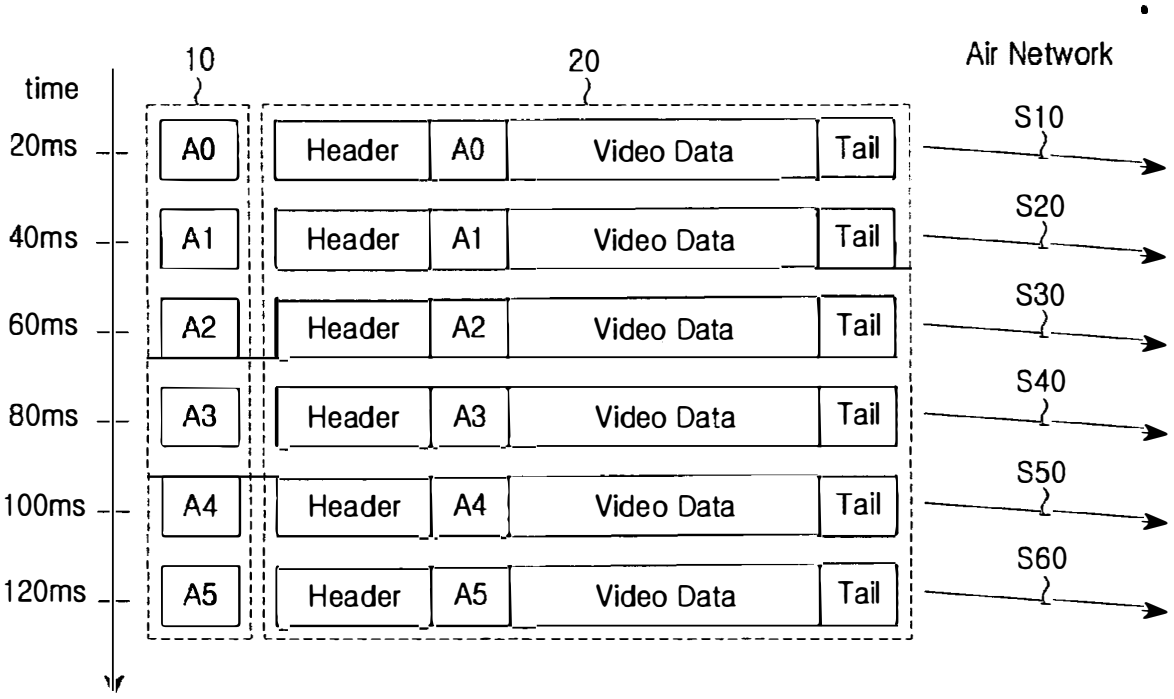

time
20ms
40ms
60ms
80ms
100ms
120ms
10
20
Air Network
A0
A1
A2
A3
A4
A5
Header
Video Data
Tail
S10
S20
S30
S40
S50
S60

FIG. 1

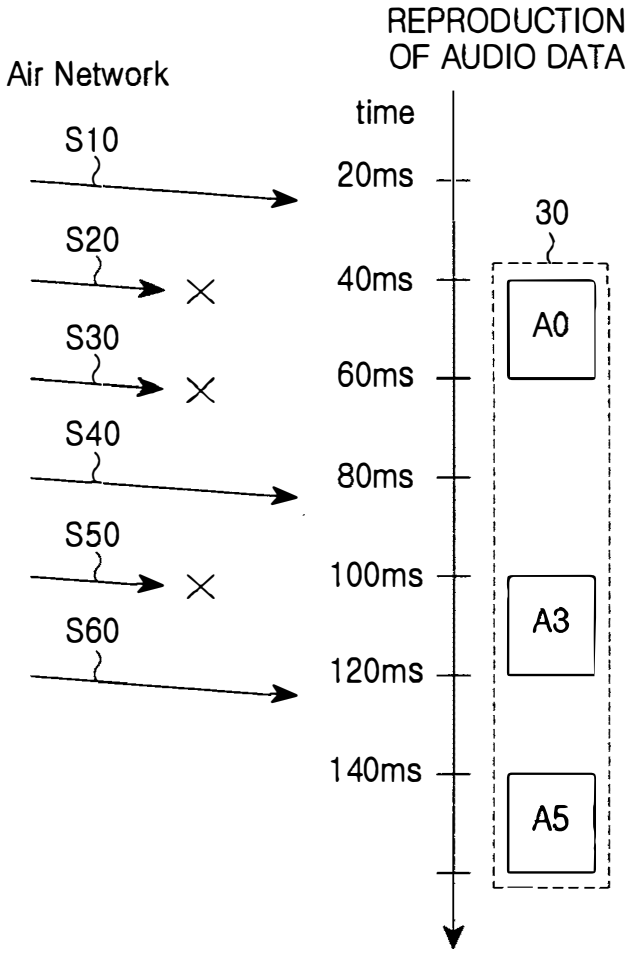

REPRODUCTION
OF AUDIO DATA
Air Network
time
S10
S20
S30
S40
S50
S60
20ms
40ms
60ms
80ms
100ms
120ms
140ms
30
A0
A3
A5

FIG.2

FIG.3A

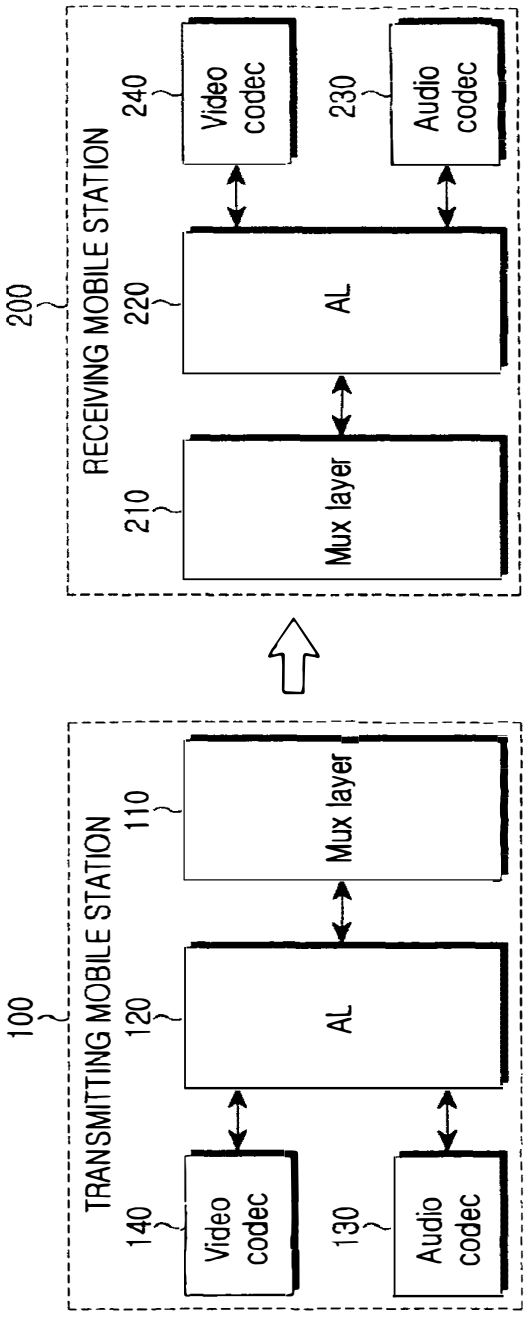
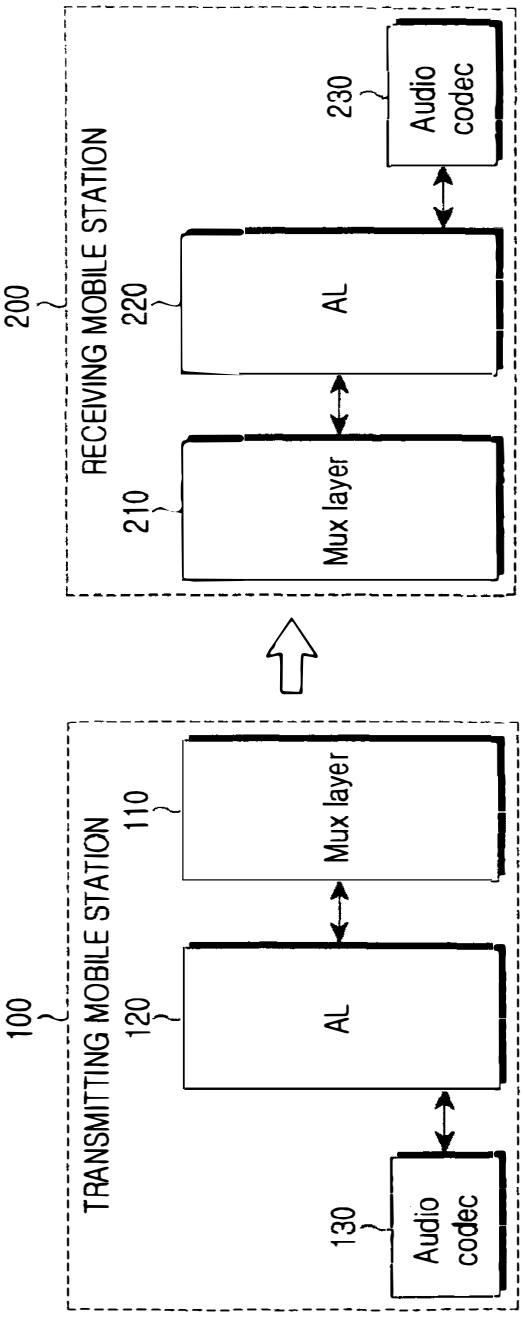

100
TRANSMITTING MOBILE STATION
110
Mux layer
120
AL
140
Video codec
130
Audio codec
200
RECEIVING MOBILE STATION
210
Mux layer
220
AL
240
Video codec
230
Audio codec

FIG.3B


100
TRANSMITTING MOBILE STATION
110
Mux layer
120
AL
130
Audio codec
200
RECEIVING MOBILE STATION
210
Mux layer
220
AL
230
Audio codec

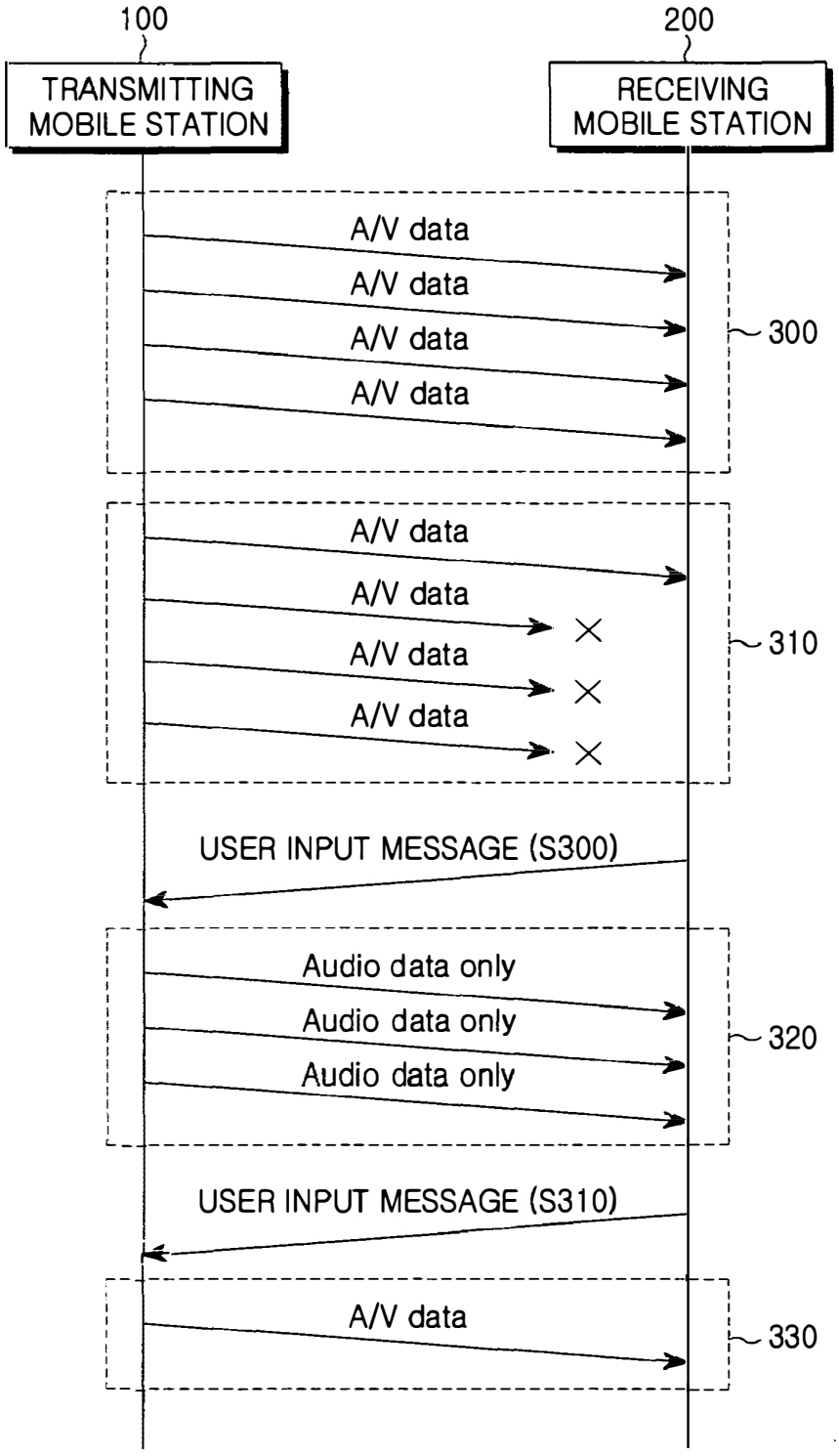

100
200
TRANSMITTING MOBILE STATION
RECEIVING MOBILE STATION
A/V data
A/V data
A/V data
A/V data
300
A/V data
A/V data
A/V data
A/V data
310
USER INPUT MESSAGE (S300)
Audio data only
Audio data only
Audio data only
320
USER INPUT MESSAGE (S310)
A/V data
330

FIG.4

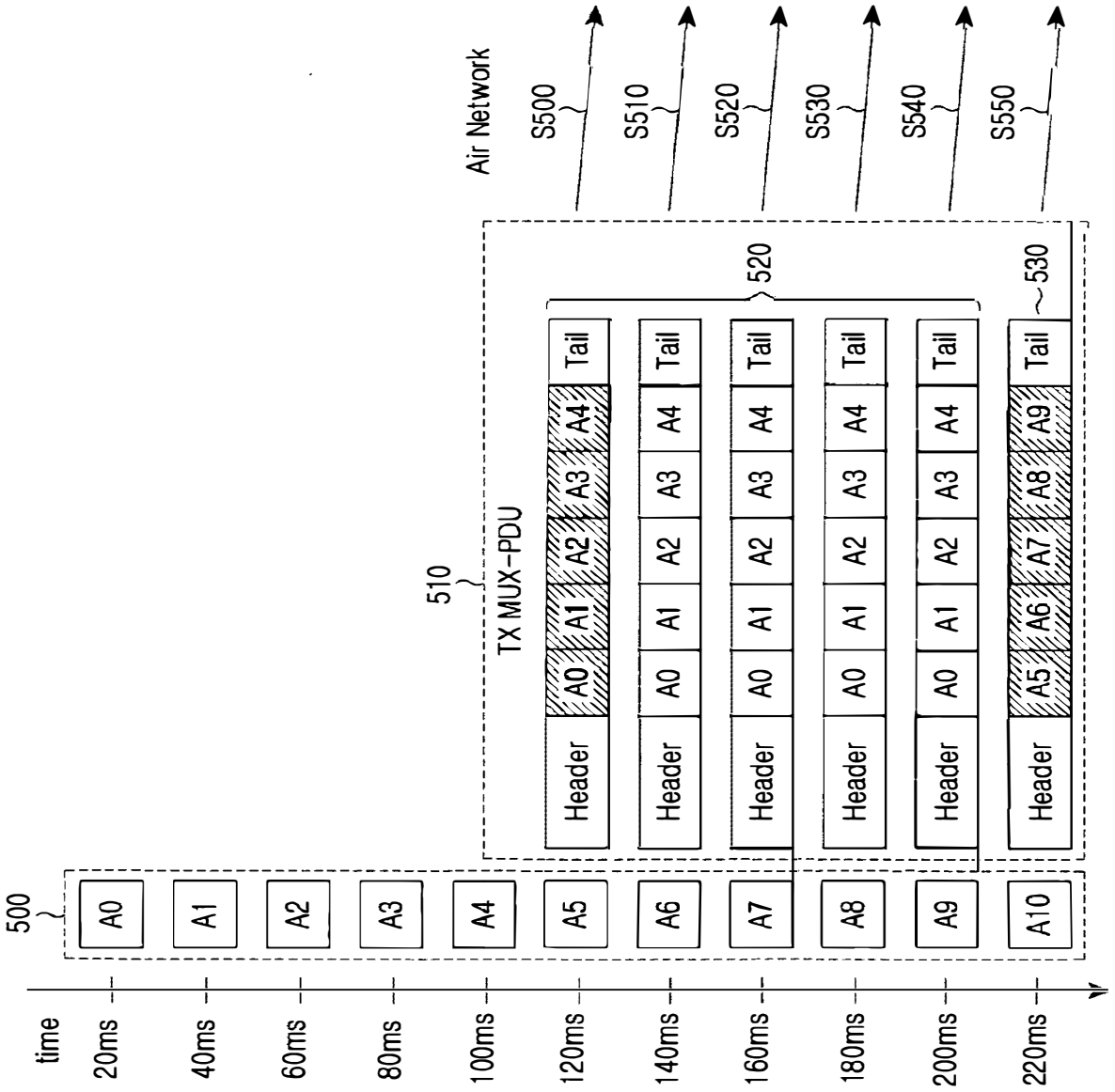

time
20ms
40ms
60ms
80ms
100ms
120ms
140ms
160ms
180ms
200ms
220ms
500
A0
A1
A2
A3
A4
A5
A6
A7
A8
A9
A10
510
TX MUX-PDU
Header
Tail
520
530
Air Network
S500
S510
S520
S530
S540
S550

FIG.5

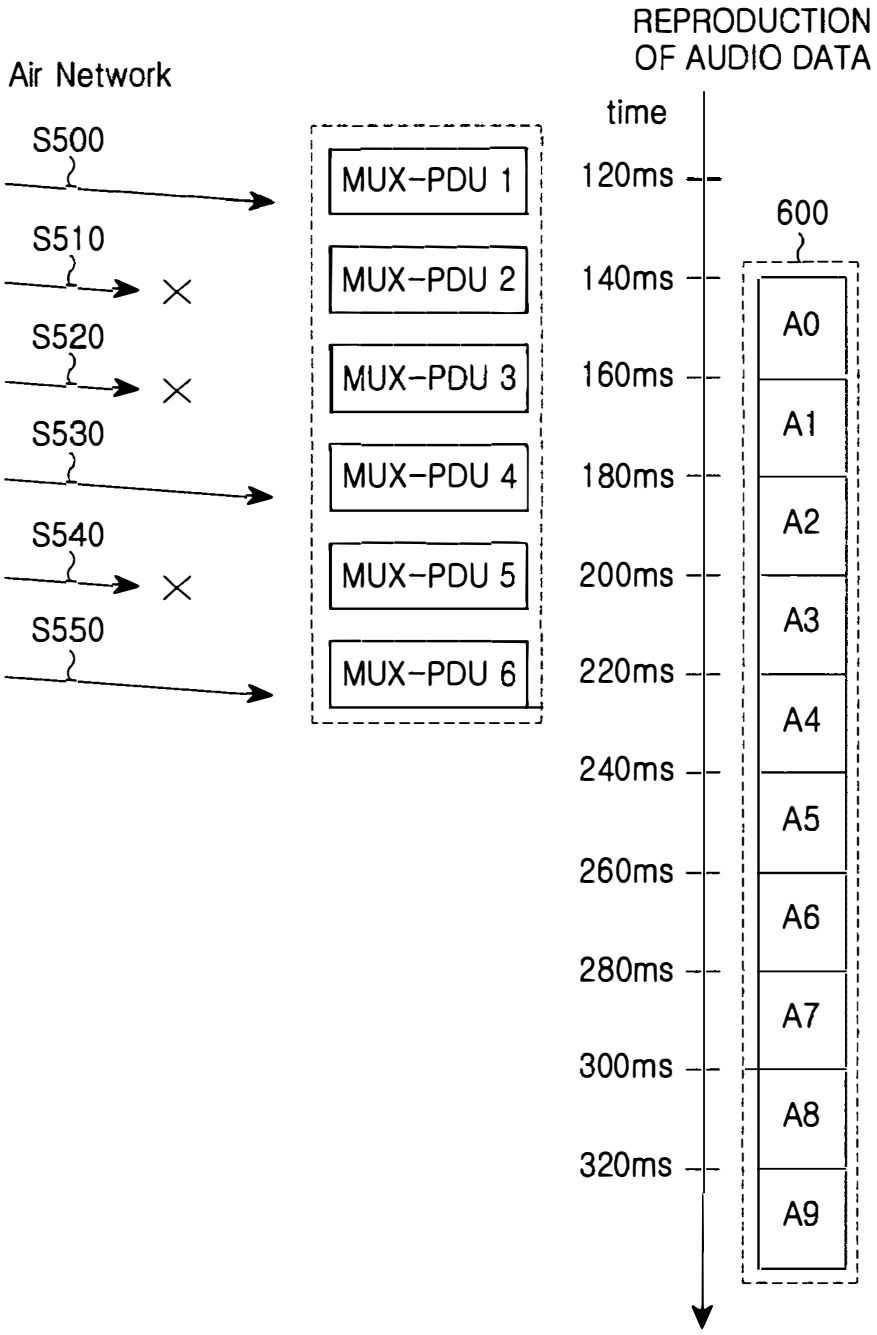

Air Network
S500
S510
S520
S530
S540
S550
MUX-PDU 1
MUX-PDU 2
MUX-PDU 3
MUX-PDU 4
MUX-PDU 5
MUX-PDU 6
REPRODUCTION OF AUDIO DATA
time
120ms
140ms
160ms
180ms
200ms
220ms
240ms
260ms
280ms
300ms
320ms
600
A0
A1
A2
A3
A4
A5
A6
A7
A8
A9

FIG.6

(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
14 September 2006 (14.09.2006)

PCT

(10) International Publication Number
WO 2006/096557 A2

(51) International Patent Classification:
*H04Q 7/20* (2006.01)

(21) International Application Number:
PCT/US2006/007713

(22) International Filing Date: 3 March 2006 (03.03.2006)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
60/658,923 4 March 2005 (04.03.2005) US

(71) Applicant *(for all designated States except US)*: **SONIM TECHNOLOGIES, INC.** [US/US]; 1875 S. Grant Street, San Mateo, California 94402 (US).

(72) Inventor; and

(75) Inventor/Applicant *(for US only)*: **GUNDU, Veerabhadra** [IN/US]; 674 Gail Avenue #G2, Sunnyvale, California 94086 (US).

(74) Agent: **IPSG, PC**; PO Box 700640, Damon K.I. Kali, San Jose, California 95170 (US).

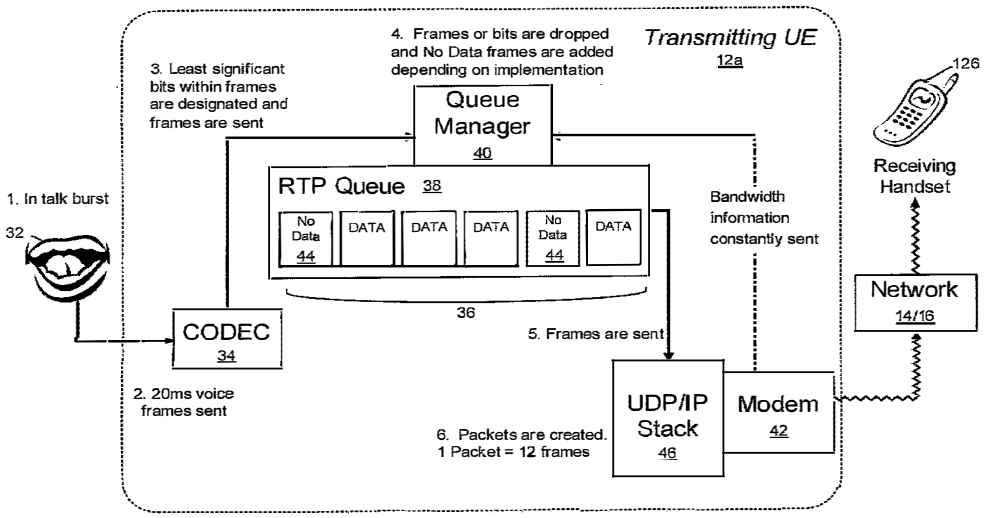
(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, LY, MA, MD, MG, MK, MN, MW, MX, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK, SL, SM, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, YU, ZA, ZM, ZW.

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

**Published:**
— *without international search report and to be republished upon receipt of that report*

*For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.*

**(54) Title:** RESTRUCTURING DATA PACKETS TO IMPROVE VOICE QUALITY AT LOW BANDWIDTH CONDITIONS IN WIRELESS NETWORKS




**(57) Abstract:** Methods and client devices for transmitting data packets based on audio input from a client device to a communications network in a communication session by receiving audio input on the client device; encoding audio input into audio frames comprising audio data indicative of the audio input; measuring bandwidth utilization of the communications network; removing a portion of the audio data if the bandwidth utilization is above a first pre-determined threshold; grouping the audio frames into data packets; and transmitting the data packets over the communications network. Alternate embodiments for removing portions of the audio data include removing at least one audio frame from each of a selected set of the plurality of data packets. Additional embodiments for removing portions of the audio data include removing a set of least significant bytes from each of a selected set of audio frames.

WO 2006/096557 A2

# RESTRUCTURING DATA PACKETS TO IMPROVE VOICE QUALITY AT LOW BANDWIDTH CONDITIONS IN WIRELESS NETWORKS

## BACKGROUND

**[Para 1]** Mobile cellular communication is evolving beyond traditional voice telephony towards more sophisticated services, such as Push-To-Talk (PTT). Similar to conventional walkie-talkie communication, PTT enables mobile communication users to send a voice message to one or more recipients over a mobile phone by simply pushing a key (*i.e.*, PTT button, etc.).

**[Para 2]** One particular version of PTT, called PTT-over-Cellular (PoC), has started to be implemented in wireless data networks such as GSM/GPRS , EDGE, UMTS and CDMA cellular networks. By using internet protocols (*i.e.*, an internet protocol network), these networks can provide a packet-based data service that enables information to be sent and received across a mobile telephone network. In addition, the use of internet protocols also facilitates PoC through the use of instant connections. That is, information can be sent or received immediately as the need arises, subject to available time slots at the air interface.

**[Para 3]** PTT, including PoC-based PTT, is half-duplex. That is, all participants typically use a single frequency or channel for both transmission and reception. Either a participant speaks or listens, but not both. This is in contrast to traditional cellular communication that is full-duplex (*e.g.*, like a regular wired phone), in which at least one channel or frequency is assigned to talk, and another separate one is assigned to listen such that both speaking and listening can occur simultaneously.

**[Para 4]** For audio/video data transmissions, PoC applications require the transmission of signaling packets using a signaling protocol, *e.g.*, Session Initiation Protocol (SIP), and data packets using a data protocol, *e.g.*, Real Time Protocol (RTP). SIP is a signaling protocol for Internet conferencing, telephony, presence, events notification, and instant messaging. RTP is an Internet-standard protocol for the transport of real-time data, including audio and video media. It can be used for media-on-demand as well as interactive services such as Internet telephony. RTP consists of a data and a control part. The latter is called Real Time Control Part (RTCP).

**[Para 5]** Currently when using PoC, there is no indication to the talking user of the quality of the outgoing voice burst. The talking user thinks his voice being received properly, until he receives a response from the listening user. Since only one user can speak at a time,

there is no way of confirming the quality of service of the system until the user releases the channel and the listener replies. Even a minor degradation of network bandwidth can cause voice quality issues. The methods of the present invention involve restructuring voice data packets in such a way as to minimize the impact of this degradation. For most companies trying to implement 3GPP/3GPP2 based voice application requirements on handsets, voice quality can be enhanced by this mechanism.

**[Para 6]** PoC may be implemented over a variety of access networks, including GPRS according to 3GPP Release 97/98, EGPRS according to 3GPP Release 99 or later releases, and UMTS according to Release 99 or later releases. For these networks, a PoC implementation preferably follows these recommendations:

**[Para 7]** • The PoC implementation should work in an access network that delivers a throughput of 7.2 kbps or more.

**[Para 8]** • The QoS (Quality of Service) profile parameters should be set such that the Radio Link Control (RLC) uses an acknowledged mode of operation.

**[Para 9]** • If streaming traffic class is supported by the access network, PoC should use this traffic class for the exchange of RTP/RTCP data.

**[Para 10]** • The POC client should support Adaptive Multi Rate (AMR) 5.15 as the mandatory and default CODEC, with optional support of AMR 4.75 being desirable. The support of any other AMR CODEC is at design discretion.

**[Para 11]** • The AMR payload format should use the octet-aligned mode (byte aligned) without interleaving and without CRCs.

**[Para 12]** If traffic class streaming can be supported in the GPRS network, then an interactive traffic class Packet Data Protocol (PDP) context is preferably used for SIP and HTTP signaling; and a streaming traffic class PDP context is preferably used for the RTP/RTCP packets. If streaming is not available, then either two interactive PDP contexts may be used (one interactive PDP context intended for PoC signaling and one interactive PDP context for RTP media), or a single PDP context may be used for both PoC signaling and RTP media.

**[Para 13]** In order to ensure optimal service quality for PoC in GPRS networks, the QoS profile parameter values are carefully selected by the user equipment (UE) in PDP context activation requests. Since 3GPP Release 97/98 compliant networks do not provide support for

a streaming traffic class, a QoS profile of a single PDP context may be shared between PoC signaling and media flows.

**[Para 14]** If using a dedicated PDP context for RTP/RTCP media, this context should be set up before or at the time of the first talk session. The RTCP traffic may be transported on the same PDP context as the SIP/HTTP signaling.

**[Para 15]** When a single PDP context is shared between media and signaling, PoC proposes some QoS parameter settings that express a compromise between satisfying different transport requirements of signaling and voice media flows to ensure the best possible overall service quality for PoC. But using traffic class streaming does not fully solve the problem. The GPRS network cannot differentiate among the various types of frames within RTP packets and the stability of multiple streams cannot be guaranteed. Also, actual bandwidth in the GPRS network can fluctuate, making scheduling and prioritization of packets important to ensure a good user experience.

**[Para 16]** Since even the best GPRS network is not able to guarantee any throughput to the UE, the PoC service quality can only be ensured if the radio access network is appropriately dimensioned. The following configurative means are available to improve the performance of the PoC service:

**[Para 17]** • Radio channels can be assigned exclusively to PS data traffic (to avoid pre-emption by CS flows).

**[Para 18]** • The maximal number of PS users multiplexed on the same timeslot (separate for UL and DL) can be limited.

**[Para 19]** • The weight assigned to the priority level (related to the Precedence Class parameter value) of the PoC flow can be augmented.

**[Para 20]** • UDP/IP header compression (RFC2507) can be configured to reduce the required radio link capacity.

**[Para 21]** If the underlying access network supports traffic class streaming, the secondary PDP context is to be used for the media (voice) flows of the PoC application. In addition, the following configurative means are available to improve the performance of the PoC service:

**[Para 22]** • UDP/IP header compression (RFC2507) or RTP/UDP/IP header compression (RFC3095) can be configured to reduce the required radio link capacity.

PCT/US06/07713

**[Para 23]** Delayed release of DL Temporary Block Flows (TBFs) and Extended TBF Mode in UL (available for 3GPP Release 4 compliant networks only) can be configured to preserve the TBF over a longer period of time.

**[Para 24]** PoC is discussed in greater detail in the following technical specifications which are incorporated by reference: PoC, Architecture, PoC Release 2.0 , V2.0.8 (2004-06); PoC, Signaling Flows – UE to Network Interface (UNI), PoC Release 2.0, V2.0.6 (2004-06); and PoC User Plane, Transport Protocols, PoC Release 2.0 , V2.0.8 (2004-06). Of note, Release 1.0 is also available from the PoC Consortium as well as an upcoming PoC standard from Open Mobile Alliance (OMA).

**[Para 25]** In summary, where PTT applications operate in a limited bandwidth environment such as cellular networks, voice quality is diminished resulting in a poor user experience regardless of the type of packet compression in use. The present invention addresses this problem.

## SUMMARY

**[Para 26]** The following presents a simplified summary of some embodiments of the invention in order to provide a basic understanding of the invention. This summary is not an extensive overview of the invention. It is not intended to identify key/critical elements of the invention or to delineate the scope of the invention. Its sole purpose is to present some embodiments of the invention in a simplified form as a prelude to the more detailed description that is presented below.

**[Para 27]** One aspect of the present invention advantageously provides for strategically minimizing the size of data packets during PTT sessions.

**[Para 28]** An exemplary embodiment of the invention includes a method for minimizing the size of data packets during a push-to-talk session by reducing the number of audio frames in the talk burst and repackaging the data packets, and a method for reducing data packet size by removing the least significant bits from the audio frames. Either method may be employed to reduce data packet size, but both methods may not be employed simultaneously. In another aspect of this embodiment is a method of determining uplink bandwidth through foru different indicators: IP Buffer length indication, QoS degradation indication from the network, RTCP receiver reports, and Packet Sent indication.

**[Para 29]** Advantages of this exemplary embodiment include an effective method for reducing the size of data packets to increase voice quality and enhance PTT user experience in low bandwidth radio conditions.

## BRIEF DESCRIPTION OF THE DRAWINGS

**[Para 30]** The present invention is illustrated by way of example, and not by way of limitation, in the figures of the accompanying drawings and in which like reference numerals refer to similar elements and in which:

**[Para 31]** Figure 1 is a block diagram of an exemplary PoC system architecture in accordance with PoC Consortium specifications;

**[Para 32]** Figure 2 is a combination block diagram and flowchart of the process flow voice data packets under control of the Queuing Manager in accordance with embodiments of the present invention;

**[Para 33]** Figure 3 is a block diagram illustrating the transmission of exemplary voice bursts over good bandwidth conditions;

**[Para 34]** Figure 4 is a block diagram illustrating the transmission of exemplary voice bursts over poor bandwidth without the benefit of the present invention;

**[Para 35]** Figure 5 is a block diagram illustrating a first method for restructuring voice data packets by puncturing RTP data packets in accordance with embodiments of the present invention;

**[Para 36]** Figure 6 is an illustrative representation of a poor bandwidth condition utilizing methods in accordance with embodiments of the present invention;

**[Para 37]** Figure 7 is a block diagram illustrating a second method for restructuring voice data packets by removing least significant bytes in accordance with embodiments of the present invention; and

**[Para 38]** Figure 8 is a block diagram illustrating the transmission of exemplary voice bursts over poor bandwidth using a second method in accordance with embodiments of the present invention.

## DETAILED DESCRIPTION

**[Para 39]** The invention is described with reference to specific architectures and protocols. Those skilled in the art will recognize that the description is for illustration and to provide the best mode of practicing the invention. The description is not meant to be limiting. For example, reference is made to SIP and RTP Protocol but other protocols can be used in the

invention. Likewise, reference is made to PoC applications, while other types of wireless transmissions including Voice Over IP (VOIP) can be used in the embodiments of the present invention.

**[Para 40]** Various embodiments are described hereinbelow, including methods and techniques. It should be kept in mind that the invention might also cover articles of manufacture that includes a computer readable medium on which computer-readable instructions for carrying out embodiments of the inventive technique are stored. The computer readable medium may include, for example, semiconductor, magnetic, opto-magnetic, optical, or other forms of computer readable medium for storing computer readable code. Further, the invention may also cover apparatuses for practicing embodiments of the invention. Such apparatus may include circuits, dedicated and/or programmable, to carry out tasks pertaining to embodiments of the invention. Examples of such apparatus include a general-purpose computer and/or a dedicated computing device when appropriately programmed and may include a combination of a computer/computing device and dedicated/programmable circuits adapted for the various tasks pertaining to embodiments of the invention.

**[Para 41]** A. Overview

**[Para 42]** The present invention is described in the exemplary context of PoC applications that use SIP signaling protocol and RTP for audio/video data transmissions. As discussed in the Background section, PoC may be implemented with or without traffic class streaming. The present invention is still beneficial when traffic class streaming is in use. With or without traffic class streaming, the PoC implementation of the preferred embodiment is applicable to an access network that delivers a throughput of 7.2 kbps or more and should support AMR 5.15 as the default CODEC. Table 1 below describes the bandwidth consumption required for AMR 5.15 with robust header compression (ROHC) and without.

**[Para 43]** An AMR-NB (Adaptive Multi Rate-Narrow Band speech CODEC) is used to compress the toll quality speech (8000 samples/second). This speech coder is mainly used for speech compression in 3rd generation mobile telephony. This CODEC has eight basic bit rates, 12.2, 10.2, 7.95, 7.40, 6.70, 5.90, 5.15, and 4.75 Kbit/s. This CODEC works on the principle of Algebraic Code Excited Linear Prediction (ACELP) for all bit rates. To reduce average bit rate, this CODEC supports the discontinuous transmission (DTX), using Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) algorithms. The eight AMR CODEC bit-rates (modes) are denoted with indices 0 to 7 where 0 maps to 4.75 kbit/s mode and 7 maps to 12.2 kbit/s mode.

**[Para 44]** AMR is discussed in greater detail in the following technical specifications: TS 26.090: "AMR Speech CODEC; Speech Transcoding Functions", TS 26.093: "AMR Speech CODEC; Source Controlled Rate Operations", and TS 26.092: "AMR Speech CODEC; Comfort Noise Aspects."

| Number of frames per RTP packet | Bandwidth consumption [kbps] AMR5.15, IPv4, No ROHC | Bandwidth consumption [kbps] AMR5.15, IPv4, ROHC |
|---|---|---|
| 1 | 22.0 | 7.2 |
| 2 | 13.8 | 6.4 |
| 3 | 11.1 | 6.1 |
| 4 | 9.7 | 6.0 |
| 6 | 8.3 | 5.9 |
| 8 | 7.7 | 5.8 |
| 12 | 7.0 | 5.7 |
| 16 | 6.6 | 5.7 |

Table 1: Bandwidth Consumption for AMR5.15

**[Para 45]** Table 1 displays the number of frames per packet for the various bandwidth amounts for the AMR5.15 CODEC with and without ROHC. As shown above, in most cases, wireless systems will put 12 to 16 frames per RTP packet for a throughput of 7.2 kbps (minimum required by PoC) without ROHC compression, but there may be as few as 1 frame per RTP packet for the same throughput if using ROHC compression. This specification uses the example of 12 frames per RTP packet in describing the present invention as this represents the most widely used setting.

**[Para 46]** The PoC system establishes the AMR RTP payload attributes and mode-set when the PTT session is created. This determines how many frames will actually be packaged into each RTP packet during the PTT session. The PoC system preferably supports the default CODEC, AMR5.15 and also other AMR modes if possible. The AMR modes are one of the eight AMR CODEC bit-rates denoted with indices 0 to 7 where 0 maps to 4.75 kbit/s mode and 7 maps to 12.2 kbit/s mode. The mode-set may be re-negotiated during a PTT session. This allows a change in the number of frames per RTP packet if more bandwidth becomes available. The AMR payload format should use the octet-aligned mode (byte aligned) without interleaving and without CRCs. A Cyclic Redundancy Check (CRC) is used to protect Frames. The transmitter appends an extra n-bit sequence to every frame called a Frame Check

Sequence (FCS). The FCS holds redundant information about the frame that helps the transmitter detect errors in the frame.

**[Para 47]** The AMR parameters that are negotiated in the PTT session establishment are mode-set, ptime, maxptime, and octet-aligned. The 'maxptime' parameter, which is signaled by the PoC client device, user equipment (UE), indicates the maximum amount of media that can be encapsulated in a payload packet and is expressed as time in milliseconds. The 'maxptime' value takes into account any network delays. After SDP negotiation, the decoding the UE is able to unpack RTP packets containing any number of frames up to 'maxptime'.

**[Para 48]** The amount of media that is encapsulated in a payload packet is signaled by the 'ptime' value. The value is determined by the number of frames per RTP packet multiplied by 20ms per frame to give the interval in milliseconds that represents the amount of media which can be encapsulated in an RTP payload packet. During the talk session, the UE is able to accept SDP re-negotiations of 'ptime' up to the negotiated 'maxptime'. The encoding UE may pack fewer frames into the last RTP packet of the talk burst, regardless of what has been defined during session negotiation or adaptation.

**[Para 49]** The AMR CODEC mode used for encoding each frame is signaled with the Frame Type (FT) index in the payload table of contents. Below, Table 2 defines the various Frame Types found in RTP packets.

PCT/US2006/007713

Most Likely Choice for these Networks

| Frame Type | Mode Indication | Mode Request | Frame content (AMR mode, comfort noise, or other) | # of bits in Frame Type | # of Bits in AMR Core Frame | # of Bits in Bit Stuffing | # of octets |
|---|---|---|---|---|---|---|---|
| 0 | 0 | 0 | AMR 4.75 kbit/s | 4 | 95 | 5 | 13 |
| 1 | 1 | 1 | AMR 5.15 kbit/s | 4 | 103 | 5 | 14 |
| 2 | 2 | 2 | AMR 5.90 kbit/s | 4 | 118 | 6 | 16 |
| 3 | 3 | 3 | AMR 6.70 kbit/s (PDC-EFR) | 4 | 134 | 6 | 18 |
| 4 | 4 | 4 | AMR 7.40 kbit/s (TDMA-EFR) | 4 | 148 | 0 | 19 |
| 5 | 5 | 5 | AMR 7.95 kbit/s | 4 | 159 | 5 | 21 |
| 6 | 6 | 6 | AMR 10.2 kbit/s | 4 | 204 | 0 | 26 |
| 7 | 7 | 7 | AMR 12.2 kbit/s (GSM-EFR) | 4 | 244 | 0 | 31 |
| 8 | - | - | AMR SID | 4 | 39 | 5 | 6 |
| 9 | - | - | GSM-EFR SID | 4 | 43 | 1 | 6 |
| 10 | - | - | TDMA-EFR SID | 4 | 38 | 6 | 6 |
| 11 | - | - | PDC-EFR SID | 4 | 37 | 7 | 6 |
| 12-14 | - | - | For future use | - | - | - | - |
| 15 | - | - | No Data (No transmission/No reception) | 4 | 0 | 4 | 1 |

GPRS

EDGE

3G

Silence Frames

No Data Frame

Table 2: Frame Types

**[Para 50]** In the table above, the Frame Types 0 to 7 are the frame types for speech bits and Frame Type 15 is a No Data frame. Different networks will use different Frame Types. For example, a GPRS network is likely to use Frame Type 1, an Edge network is likely to use Frame Type 3 or 4, and a 3G network is likely to use Frame Type 7.

**[Para 51]** The methods of the present invention assume that uplink bandwidth is being monitored by the modem and being reported by the modem to the PoC Client. There are four possible standard real-time indicators of uplink bandwidth strength currently used by those proficient in the art:

**[Para 52]** • IP Buffer Length indication – for each RTP packet transmission, the platform will send information about the current IP Buffer length. For each packet transmission, a comparison is made between the current

PCT/US06/07713

length value and the previous length value to gain a measurement of the uplink bandwidth.

**[Para 53]** • QoS degradation indication – if network conditions degrade, the server can degrade the QoS for the PTT application. The UE uses the current QoS to determine the number of frames to include in each packet. The QoS value will be reevaluated by the server on a regular basis and changed as needed.

**[Para 54]** • Packet Sent indication – the UE tags each packet and stores the information in memory. When the modem sends the packet, it also sends a notification back to the client, which includes a timestamp. The client can calculate bandwidth based on the packet size and the time it took to get the packet sent and alter packet size accordingly.

**[Para 55]** • RTCP Receiver Reports – the PTT server or receiving UE may send back a report to the transmitting UE on how many packets that were received over a certain time span, e.g. audio (or voice) burst.

**[Para 56]** When any of these bandwidth indictors are implemented, the modem will send bandwidth measurement data to the UE. This data is used to determine how large the RTP packets shall be when using one of the two methods discusses in this specification.

**[Para 57]** In summary, where PTT applications operate in a limited bandwidth environment such as cellular networks, voice quality is diminished resulting in a poor user experience regardless of the type of packet compression in use. The present invention addresses the problem through two distinct methods. Either method may be implemented, but they cannot be implemented together. The first method involves selectively dropping audio frames and repackaging the data packets, allowing the receiving UE to extrapolate the contents of the missing frames. The second method involves dropping the least significant bits from the audio frames. This second method only applies to AMR CODECs while the first method applies to all CODECs.

**[Para 58]** B. Architecture

**[Para 59]** Figure 1 illustrates the system 10 of the preferred embodiment as well as the interfaces for messages transmitted between the UE 12 and various components of the PoC system 14. The system 10 includes UE 12, access network 16, Over the Air Provisioning (OTAP) server 18, IMS Core 20, PoC servers 22, and remote PoC networks 26. Access network 16 is the communications network for connecting UE 12 to the PoC system 14. In the

case of a PoC system 14 (*i.e.*, PTT-over-cellular), the access network 16 is a cellular network. The OTAP server 18 performs the following functions that are needed in support of the PoC service: provides all the needed configuration parameters from the service provider access network 16 for a PoC client (*i.e.*, UE 12), and sends a WAP-push/SMS containing a binary coded XML to every client UE 12 with default factory and network settings.

**[Para 60]** The PoC services 26 include Group List Management Server (GLMS) 28, PoC server 22, and Presence Server 30. As would be obvious to those of ordinary skill in the art, the PoC services 26 may be implemented in a single physical server, in multiple physical servers for each function, or any combination thereof.

**[Para 61]** C. Data Packet Size Reduction Process

**[Para 62]** Figure 2 is a combination block diagram and flowchart of the process flow voice data packets under control of the Queuing Manager 40 in accordance with embodiments of the present invention. When user 32 speaks, audio input is received by transmitting UE 12a and is initially sent to CODEC 34 as audio frames having a 20ms duration. CODEC 34 designates which bits within the frames 36 are the least significant bits and sends the frames 36 to the RTP Queue 38. Queue Manager 40 analyzes the contents of the RTP queue 38 according to the method being implemented (*i.e.*, frame removal or byte removal) in preparation for transmitting the frames. In addition, Queue Manager 40 receives constant information from the modem 42 as to current bandwidth conditions.

**[Para 63]** Frame Removal

**[Para 64]** If a voice frame removal method is implemented, Queue Manager 40 selects which audio (or voice) frames 36 to remove based on the uplink bandwidth utilization conditions received from modem 42. The poorer the network conditions, the more audio frames are dropped. In one embodiment, CODEC 34 has a 3-frame memory built into it that can be utilized by the Queue Manager 40 when determining how often to drop a frame. The Queue Manager 40 then inserts "No Data" frames 44 into vacated slots. Queue Manager 40 then sends the audio frames to UDP/IP Stack 46 for grouping into data packets. In this example, fewer data packets may be sent due to the reduction in audio frames. Data packets may then be sent over network 14/16 (PoC system 14 and access network 16) to receiving handset 126. In one embodiment, one data packet is comprised of a group of 12 audio frames.

**[Para 65]** Byte Removal

**[Para 66]** If the method being implemented involves removing the least significant bits from the audio (or voice) frames 36, the Queue Manager 40 removes those bits. This shrinks

the audio frame size resulting in a correspondingly smaller data packet size. Queue Manager 40 then sends the audio frames to UDP/IP Stack 46 for grouping into data packets. In this example, smaller data packets may be sent due to the reduction in audio frame size. Data packets may then be sent over network 14/16 (PoC system 14 and access network 16) to receiving handset 126. In one embodiment, one data packet is comprised of a group of 12 audio frames.

**[Para 67]** Figure 3 is a block diagram illustrating the transmission of exemplary voice bursts over good bandwidth utilization conditions. Audio (or voice) frames 36a correspond to original analog wave 50a emitted by a user (*i.e.*, user 32, Figure 2). Original analog wave 50a is digitized by a CODEC to create corresponding audio frames 36a. Any CODEC well-known in the art may be utilized without departing from the present invention. Audio frames 36a may then be grouped into communication data packets such as RTP packets 48 (*e.g.* P1-P5) by, for example, a UDP/IP Stack as described in Figure 2. RTP Packets 48 may then be transmitted from UE 12a over network 14/16 to a receiving UE 12b. Data packets are then ungrouped into audio frames 36b whereupon audio frames 36b may be decoded by a CODEC to be heard by a listener as transmitted analog wave 50b. Under good bandwidth utilization conditions, minimal distortion may be expected between original analog wave 50a and transmitted analog wave 50b.

**[Para 68]** Figure 4 is a block diagram illustrating the transmission of exemplary voice bursts over poor bandwidth utilization conditions without the benefit of the present invention. In this example, data packet P2 is lost in transmission over the network 14/16. It may be appreciated that data packets may be lost for any number of reasons well-known in the art. When a data packet is lost, a listener may experience a gap in audio (or voice burst) resulting in a poor user experience. Transmitted audio wave 50b illustrates the loss as a precipitous drop, followed by a zero state, which is in turn followed by a precipitous rise in the audio wave. In contrast, methods of the present invention may be utilized to enhance user experience by reorganizing data packets 48 so that complete data packets are never dropped even under poor network conditions.

**[Para 69]** One method for improving audio transmission under poor bandwidth conditions involves removing audio frames and repackaging data packets in such a way as to spread the loss of audio frames across one or more data packets in an audio (or voice) burst. Figure 5 is a block diagram illustrating a first method for restructuring data packets by puncturing RTP data packets in accordance with embodiments of the present invention. This method utilizes IP buffer utilization as the metric for deciding whether available bandwidth is scarce and, as such,

whether audio frame dropping is necessary. IP buffer utilization may be high for any number of reasons including poor network conditions or competing high-priority applications on same device, for example. In order to achieve a smooth reaction curve and to avoid removing audio frames unnecessarily, a sliding window may be utilized that includes a predefined number of data packet measurement points for the IP buffer to calculate an average value before triggering an audio frame reduction. For example, RTP Sample x (502) may be utilized to calculate a first average value. Thereafter, RTP Sample x + 1 (504) may be utilized to calculate a second average value. The method may continue for any number of samples. By utilizing overlapping data packets in the calculation, the method may create a smoother reaction curve. Furthermore, the method can be made more responsive to change by utilizing fewer samples or less responsive to change by utilizing more samples.

**[Para 70]** Still further, a number of pre-determined thresholds may be defined. For example, a first pre-determined threshold may be configured to respond to 60% IP buffer utilization by removing one audio frame from each packet before transmission 506. In another example, a second pre-determined threshold may be configured to respond to 80% IP buffer utilization by removing two audio frames from each packet before transmission 508. Further, at an 80% IP buffer utilization level, Queue Manager 40 may provide a low-bandwidth notification message to inform the user that a critical bandwidth condition exists. In some embodiments, a temporal element, such as a pre-determined time period, may also be configured. That is, if a bandwidth utilization condition exists for longer than a specified pre-determined time period (*e.g.* z = 3 seconds) the user may be prompted to decide whether to continue a talk burst 508. If, in another example, the IP buffer reaches a third pre-determined threshold such as 100% (*i.e.* full-utilization) 510, then a flow-off timer in Queue Manager 40 may be initiated wherein Queue Manager 40 will not release any more data packets 36 from RTP Queue 38 until the flow-off timer expires or the 100% utilization condition no longer exists. Thus, no additional data packets are sent to the IP buffer in UDP/IP stack 46 while a 100% bandwidth utilization condition exists. In still another example, where an IP buffer maintains a level of 100% for up to a specified pre-determined time period such as, for example, 3 seconds, a notification message may be provided by Queue Manager 40 recommending that the user cease talking and release the floor 510. It may be appreciated that the several bandwidth utilization parameters provided are for illustrative purposes only. As such, any number of parameters may be selected without departing from the present invention.

**[Para 71]** Figure 6 is an illustrative representation of a poor bandwidth utilization condition utilizing methods in accordance with embodiments of the present invention. As

illustrated, transmitted audio wave 50b more closely resembles original audio wave 50a as compared with audio wave depicted in Figure 4. In this example, every 5th audio frame is dropped before the remaining audio frames are packaged into data packets. That is, in the embodiment illustrated, the method determines that poor network conditions exist that may result in a lost data packet (*e.g.* data packet P2, Figure 4) whereupon some number of audio frames may be dropped and remaining audio frames are grouped in a manner that avoids a complete data packet loss having a contiguous group of original audio frames. Instead, a distributed portion of the original audio frames are removed thus yielding a more desirable result. In many examples, the loss of original audio frames 36a may not be readily discernable to a listener. IT may be appreciated that the shape of transmitted audio wave 50b more closely resembles the shape of original audio wave 50a than in the scenario depicted in Figure 4. Thus a listener may hear a smoother audio (or voice burst) without obvious gaps.

**[Para 72]** In a similar manner, an embodiment may also be applied when using the QoS degradation indication as received from 3GPP Release 99 networks as input, when using RTCP Receiver Reports as input, or when using the packet sent indication as input. All of these data measurements can be utilized in a graph such as depicted in Figure 5, where kbps replaces IP buffer utilization as the metric on the y-axis. In the case of packet sent indication, the throughput in kbps for every packet 48 is calculated by taking the size of the packet 48 and dividing it by the time it took to travel to the modem 42 and back again.

**[Para 73]** Figure 7 is a block diagram illustrating a second method for restructuring audio data packets by removing a set of least significant bytes in accordance with embodiments of the present invention. The method includes reducing each audio frame size by removing the Least Significant Bytes (LSB) from each audio frame 36. The triggering of this method is similar to that shown in Figure 5, *i.e.* thresholds are set vis-à-vis the IP buffer utilization or measured kbps throughput. The difference lies in the method by which bandwidth utilization is accounted for.

**[Para 74]** In case of Adaptive Multi Rate (AMR), transmitting CODEC 34a (acting as an encoder) on a transmitting UE 12a marks and distinguishes between Most Significant Bytes (MSB) 52, Ordinary Bytes 54, and Least Significant Bytes (LSB) 56 based on captured voice samples. As such, the transmitting Queue Manager 40a may utilize this information to remove LSB 56 thus creating a new payload consisting of 9 bytes/frame instead of 13 bytes/frame. LSB may include, for example, off frequency signals that do not contribute to speech clarity. The 9 byte/frame payload 60b is transferred across the network 14/16 to the receiving UE 12b. Once receiving Queue Manager 40b receives the 9 byte/frame payload 60b, Queue Manager

40b augments the frame with dummy bits 58 in order to recreate a standard 4.75 kbps AMR Payload Type 60c having 13 bytes/frame. The recreated 13 bytes/frame payload 60c may then be processed by the receiving CODEC 34b (acting as a decoder). In this manner, a smaller data packet may be sent over the network 14/16 while preserving audio integrity. The resulting audio burst quality, as illustrated in figure 8 by transmitted audio wave 50b more nearly resembles original audio wave 50a. Thus, audio transmissions under poor bandwidth utilization conditions may be achieved that similar to audio transmissions under good bandwidth utilization conditions as illustrated in Figure 3.

**[Para 75]** D. Conclusion

**[Para 76]** While this invention has been described in terms of several embodiments, there are alterations, permutations, and equivalents, which fall within the scope of this invention. It should also be noted that there are many alternative ways of implementing the methods and apparatuses of the present invention. It is therefore intended that the following appended claims be interpreted as including all such alterations, permutations, and equivalents as fall within the true spirit and scope of the present invention.

CLAIMS

What is claimed is:

1. A method for transmitting a plurality of data packets based on an audio input from a first client device to a communications network in a communication session, comprising the steps of:

receiving the audio input on the first client device;

encoding the audio input into audio frames comprising audio data indicative of the audio input;

measuring a bandwidth utilization of the communications network;

removing a portion of the audio data if the bandwidth utilization is above a first pre-determined threshold;

grouping the audio frames into the plurality of data packets; and

transmitting the data packets over the communications network.

2. The method of claim 1, wherein the removing the portion of the audio data comprises removing at least one audio frame from each of a selected set of the plurality of data packets.

3. The method of claim 2, wherein the removing the portion of the audio data comprises the steps of:

removing at least one of the audio frames from the selected set of the plurality of data packets if the bandwidth utilization is above the first pre-determined threshold; and

removing at least two of the audio frames from the selected set of the plurality of data packets if the bandwidth utilization is above a second pre-determined threshold.

4. The method of claim 3 further comprising:

creating a first notification message if the bandwidth utilization is above the second pre-determined threshold;

creating a second notification message if the bandwidth utilization is above the second pre-determined threshold for a pre-determined time period;

terminating transmission of the plurality of data packets if the bandwidth utilization is above a third pre-determined threshold; and

creating a third notification message if the bandwidth utilization is above the third pre-determined threshold.

5. The method of claim 1, wherein the removing the portion of the audio data comprises removing a set of least significant bytes from each of a selected set of audio frames.

6. The method of claim 5 further comprising:

creating a first notification message if the bandwidth utilization is above a second pre-determined threshold;

creating a second notification message if the bandwidth utilization is above the second pre-determined threshold for a pre-determined time period;

terminating transmission of the plurality of data packets if the bandwidth utilization is above a third pre-determined threshold; and

creating a third notification message if the bandwidth utilization is above the third pre-determined threshold.

7. The method recited in claim 1, wherein the bandwidth utilization is measured based upon an indicator selected from the group consisting of: IP buffer length, QoS degradation, RTCP receiver reports, and packet sent indication data.

8. The method recited in claim 1, further comprising the steps of:

receiving the plurality of data packets from the communications network by a second client device;

decoding the plurality of data packets into the audio frames; and

reproducing the audio input based upon the audio frames on the second client device.

9. The method recited in claim 1, wherein the communication session is selected from the group consisting of: a push-to-talk session, a push-to-talk over cellular session, and a voice over IP (VOIP) session.

10. An improved client device enabled to transmit a plurality of data packets based on an audio input over a communications network during a communication session, based on a bandwidth utilization of the communications network, the improvement comprising:

means for receiving the audio input for transmission on the communications network;

means for encoding the audio input into audio frames comprising audio data indicative of the audio input;

means for measuring a bandwidth utilization of the communications network;

means for removing a portion of the audio data if the bandwidth utilization is above a first pre-determined threshold;

means for grouping the audio frames into the plurality of data packets; and

means for transmitting the plurality of data packets over the communications network.

11. The device recited in claim 10, wherein the means for removing the portion of the audio data comprises means for removing at least one audio frame from each of a selected set of the plurality of data packets

12. The device recited in claim 11, wherein the means for removing the portion of the audio data comprises:

means for removing at least one of the audio frames from the selected set of the plurality of data packets if the bandwidth utilization is above the first pre-determined threshold; and

means for removing at least two of the audio frames from the selected set of the plurality of data packets if the bandwidth utilization is above a second pre-determined threshold.

13. The device of claim 12 further comprising:

means for creating a first notification message if the bandwidth utilization is above the second pre-determined threshold;

means for creating a second notification message if the bandwidth utilization is above the second pre-determined threshold for a pre-determined time period;

means for terminating transmission of the plurality of data packets if the bandwidth utilization is above a third pre-determined threshold; and

means for creating a third notification message if the bandwidth utilization is above the third pre-determined threshold.

14. The device of claim 10, wherein the means for removing the portion of the audio data comprises means for removing a set of least significant bytes from each of a selected set of audio frames.

15. The device recited in claim 10, wherein the means for measuring bandwidth utilization measures an indicator selected from the group consisting of: IP buffer length, QoS degradation, RTCP receiver reports, and packet sent indication data.

16. The device recited in claim 10, further comprising the steps of:

means for receiving the plurality of data packets from the communications network by a second client device;

means for decoding the plurality of data packets into the audio frames; and

means for reproducing the audio input based upon the audio frames on the second client device.

17. The device recited in claim 10, wherein the communication session is selected from the group consisting of: a push-to-talk session, a push-to-talk over cellular session, and a voice over IP (VOIP) session.

18. A client device enabled to transmit a plurality of data packets based on an audio input over a communications network in a communication session comprising:

a CODEC configured to create audio frames comprising audio data representative of an audio input;

a queue coupled to the CODEC and configured to temporarily store the audio frames;

a queue manager configured to control audio frame output from the queue and further configured to remove audio data from the queue;

a UDP/IP stack configured to group the audio frames into a plurality of data packets; and

a modem configured to transmit the plurality of data packets over the communication network and to provide a bandwidth utilization indicator to the queue manager.

19. The device recited in claim 18, wherein the queue manager is further configured to remove audio data by removing audio frames from a selected set of the plurality of data packets in response to the bandwidth utilization indicator.

20. The device recited in claim 18, wherein the queue manager is further configured to remove audio data by removing at least some least significant bytes from the data packets in response to the bandwidth utilization indicator.

21. The device recited in claim 18, wherein the queue manager is further configured to stop the transmission of data packets if the bandwidth utilization indicator indicates full capacity; and to provide a notification message if the bandwidth utilization indicator indicates full capacity.

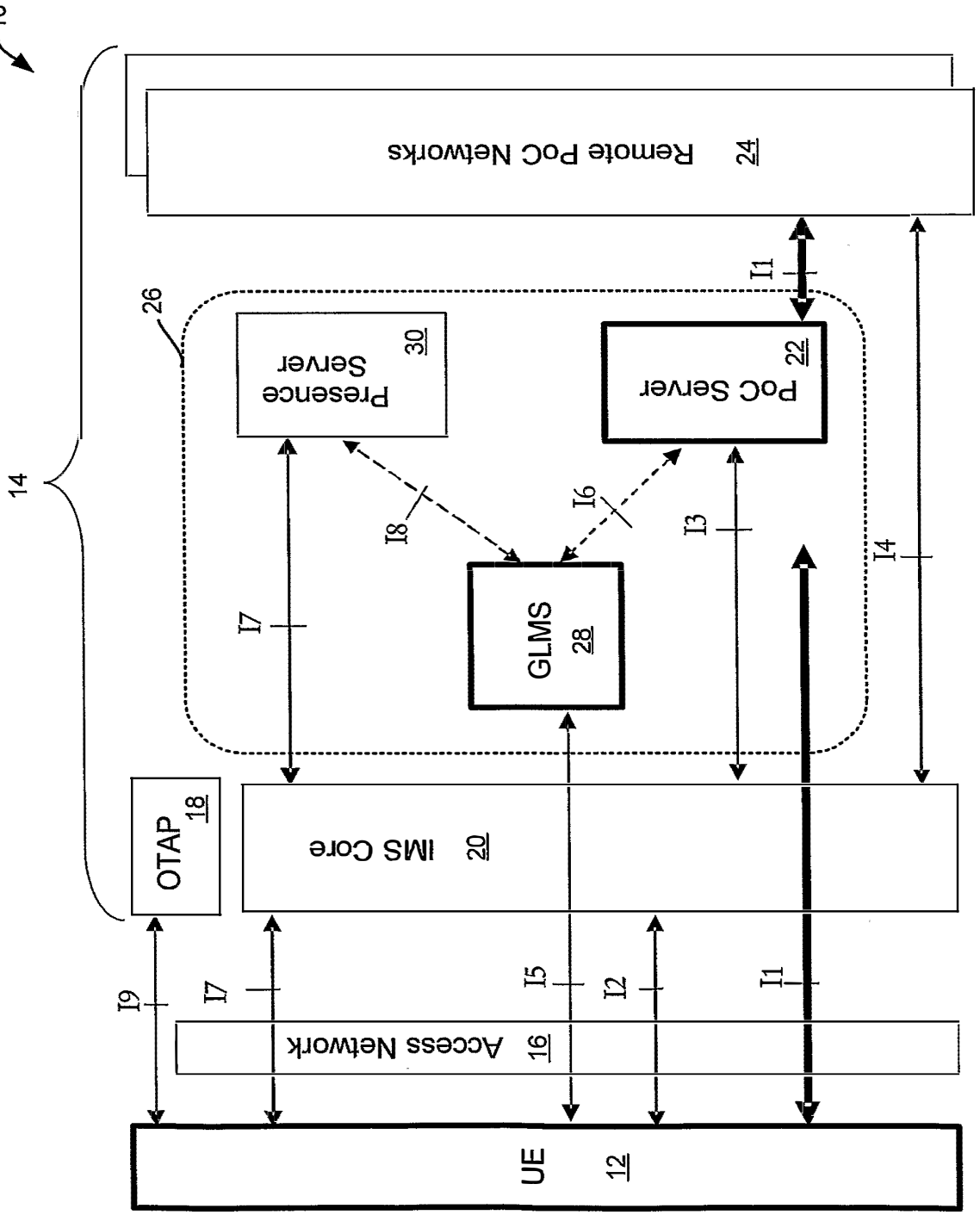

10
14
Remote PoC Networks
24
I1
26
Presence Server
30
PoC Server
22
I8
I6
I3
I4
I7
GLMS
28
OTAP
18
IMS Core
20
I9
I7
I5
I2
I1
Access Network
16
UE
12

FIG. 1

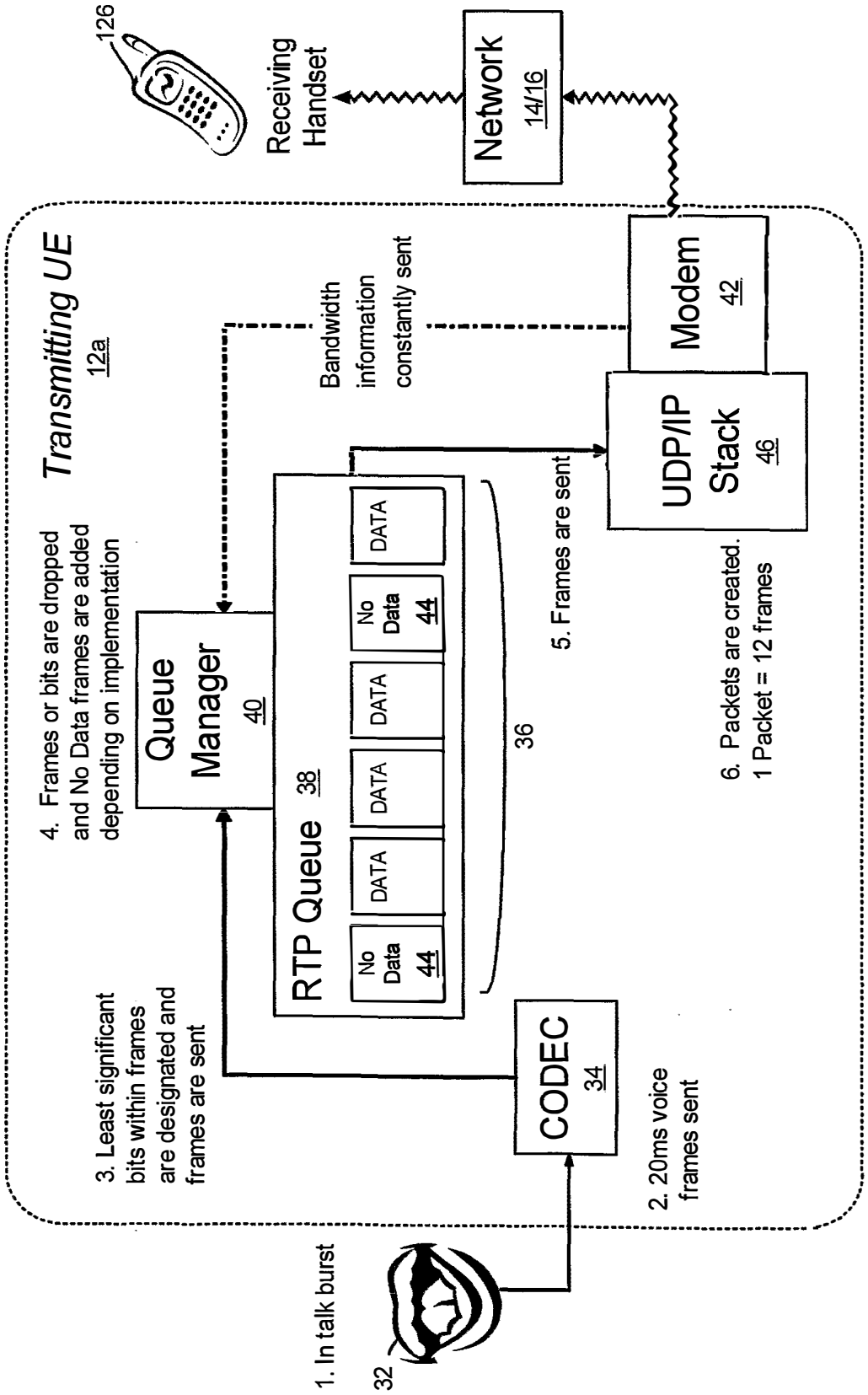

126
Receiving Handset
Network 14/16
Transmitting UE 12a
Modem 42
Bandwidth information constantly sent
UDP/IP Stack 46
5. Frames are sent
6. Packets are created. 1 Packet = 12 frames
4. Frames or bits are dropped and No Data frames are added depending on implementation
Queue Manager 40
RTP Queue 38
No Data 44
DATA
DATA
DATA
No Data 44
DATA
36
3. Least significant bits within frames are designated and frames are sent
CODEC 34
2. 20ms voice frames sent
1. In talk burst
32

FIG. 2

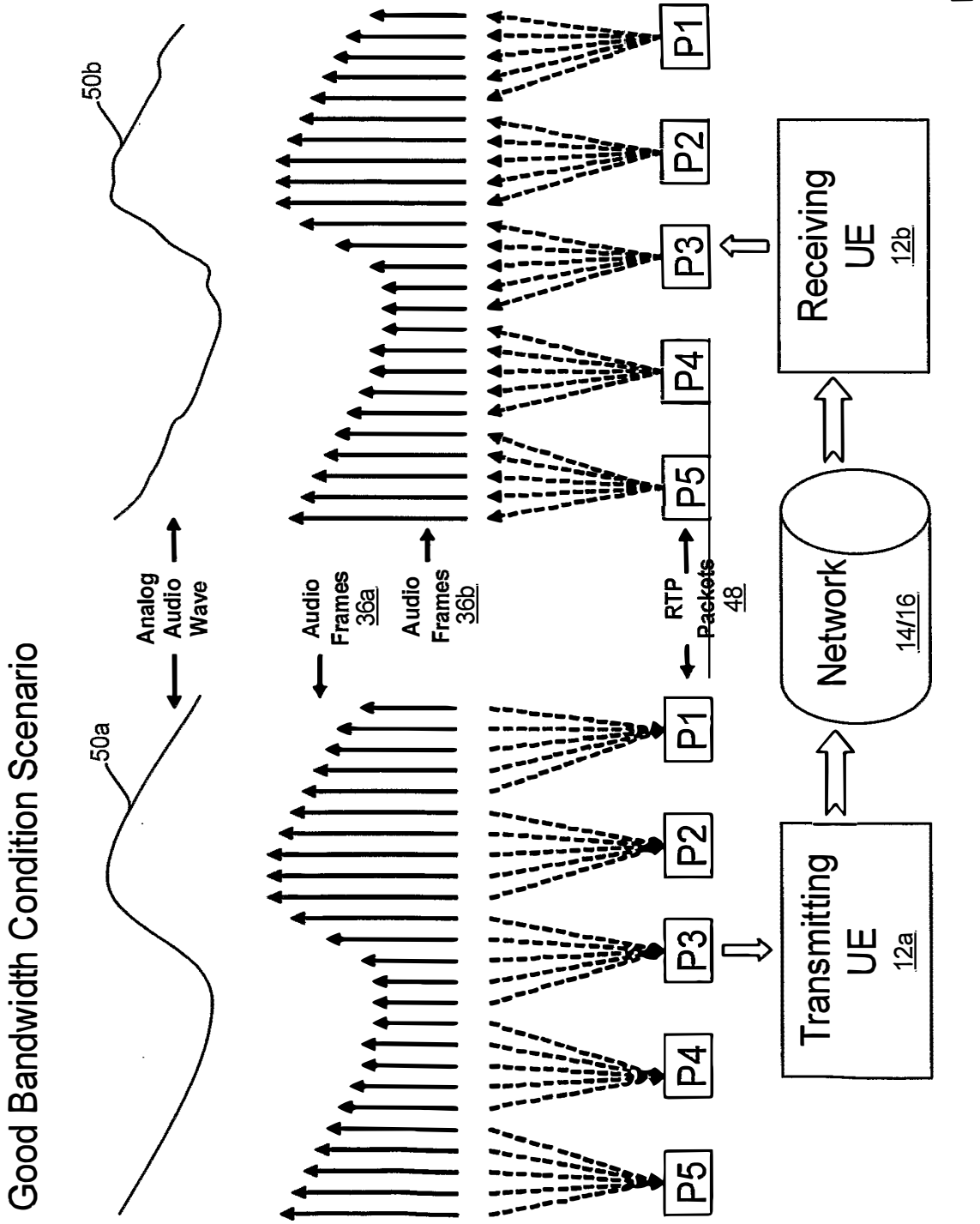

Good Bandwidth Condition Scenario
50a
50b
Analog Audio Wave
Audio Frames 36a
Audio Frames 36b
RTP Packets 48
P1
P2
P3
P4
P5
Transmitting UE 12a
Network 14/16
Receiving UE 12b

FIG. 3

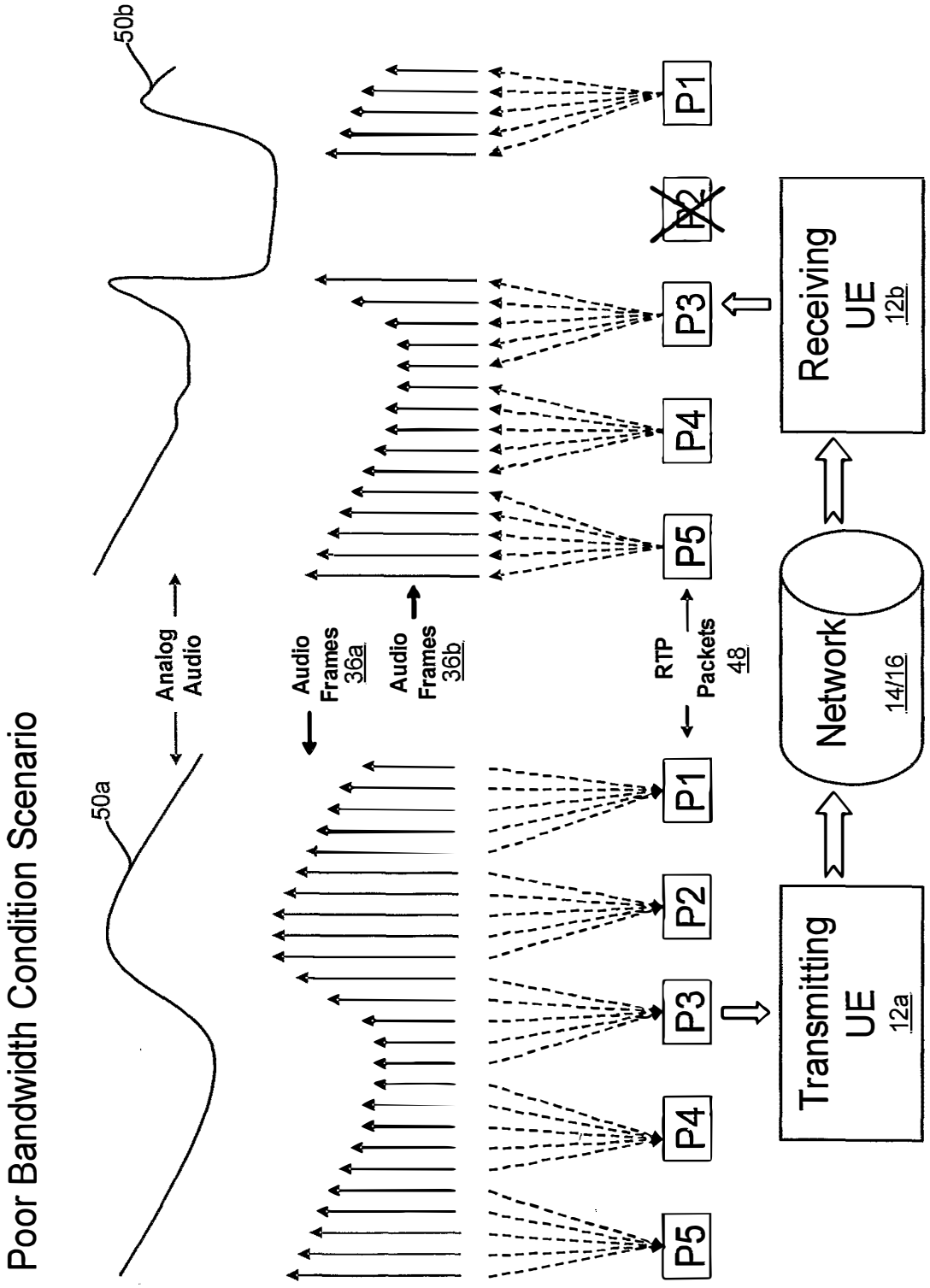

Poor Bandwidth Condition Scenario
50a
50b
Analog Audio
Audio Frames 36a
Audio Frames 36b
RTP Packets 48
P1
P2
P3
P4
P5
Transmitting UE 12a
Network 14/16
Receiving UE 12b

FIG. 4

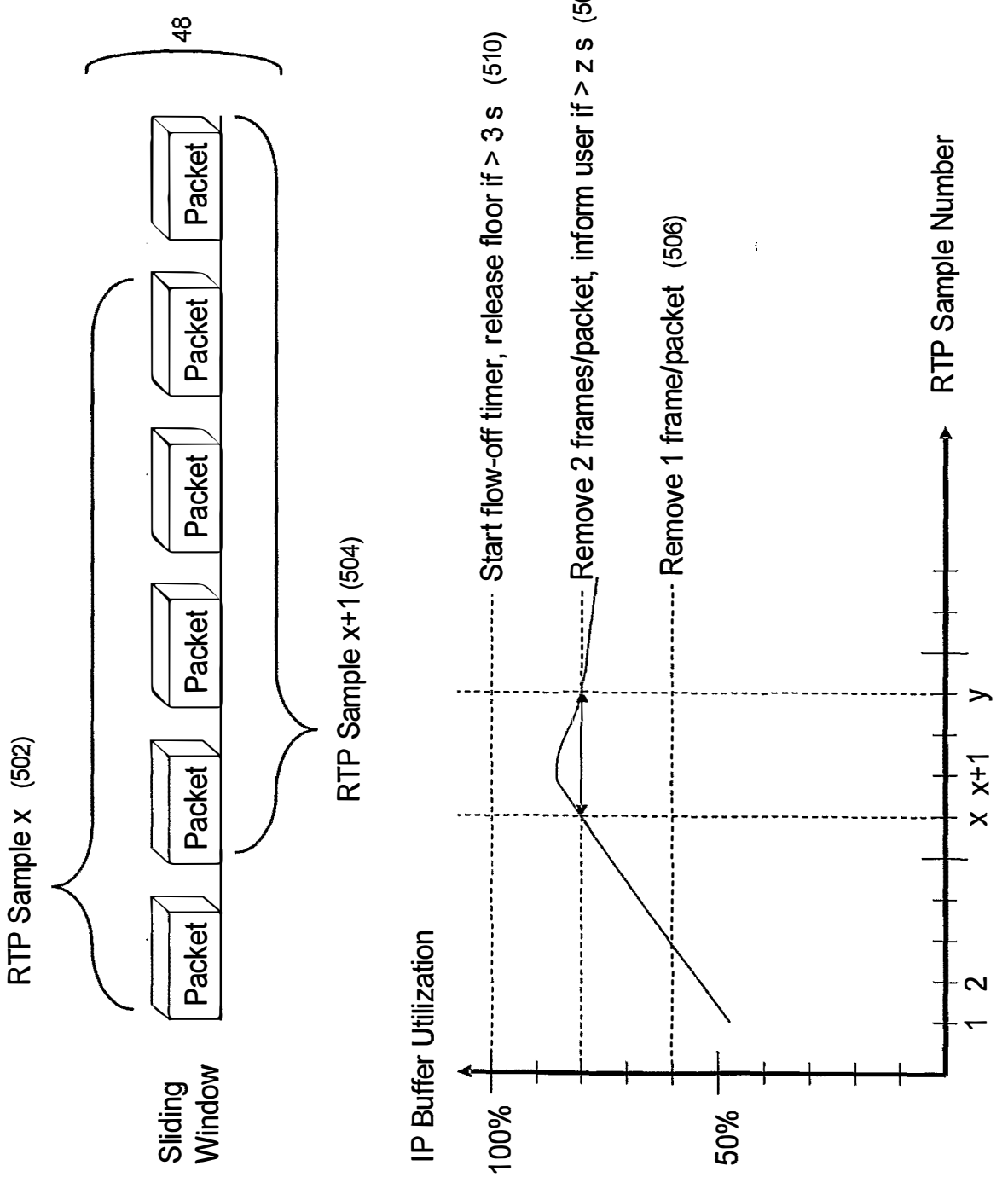

48
RTP Sample x (502)
Packet
Packet
Packet
Packet
Packet
Packet
RTP Sample x+1 (504)
Sliding Window
IP Buffer Utilization
100%
50%
Start flow-off timer, release floor if > 3 s (510)
Remove 2 frames/packet, inform user if > z s (508)
Remove 1 frame/packet (506)
1 2
x x+1
y
RTP Sample Number

FIG. 5

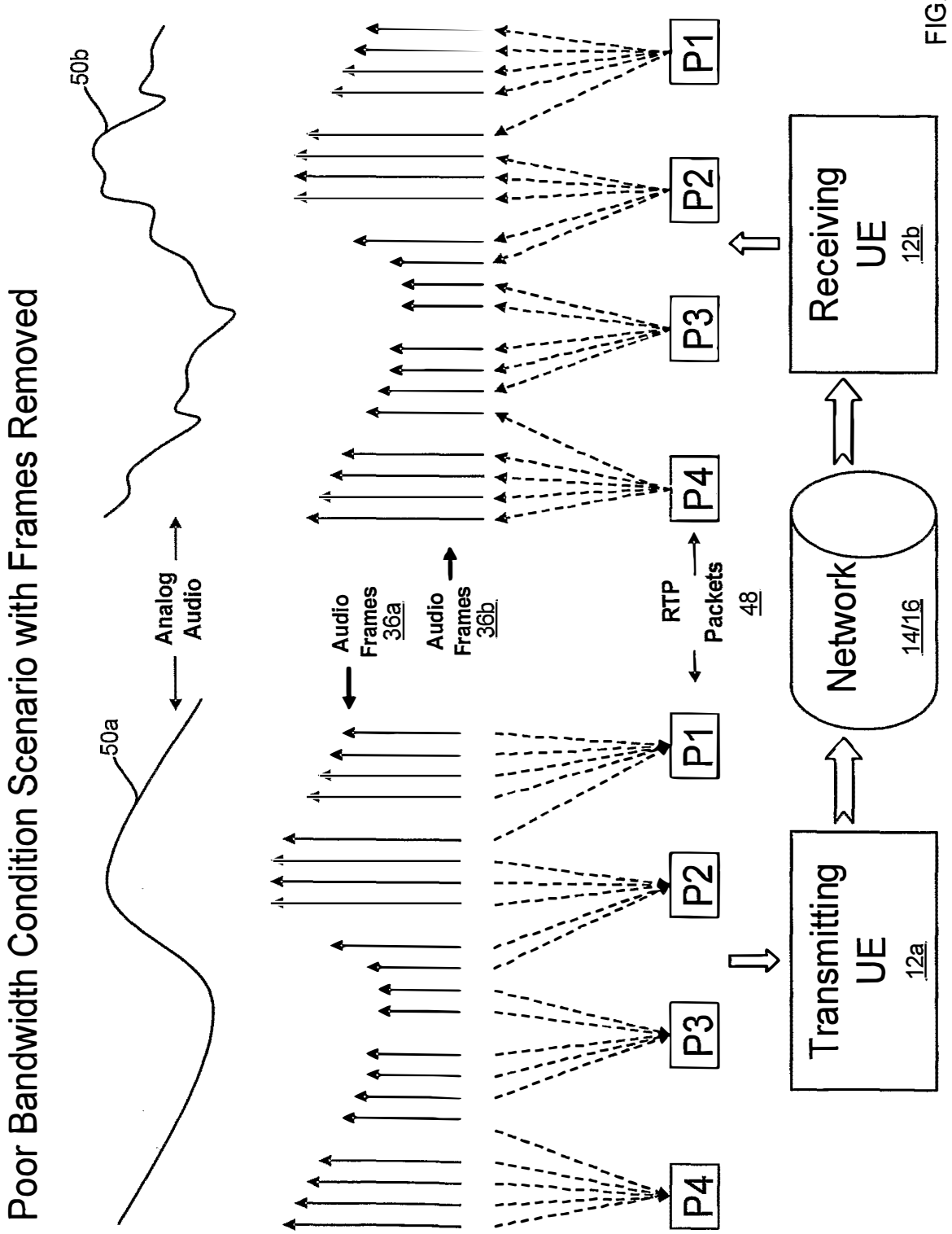

Poor Bandwidth Condition Scenario with Frames Removed
50a
50b
Analog Audio
Audio Frames 36a
Audio Frames 36b
RTP Packets 48
P1
P2
P3
P4
Transmitting UE 12a
Network 14/16
Receiving UE 12b

FIG. 6

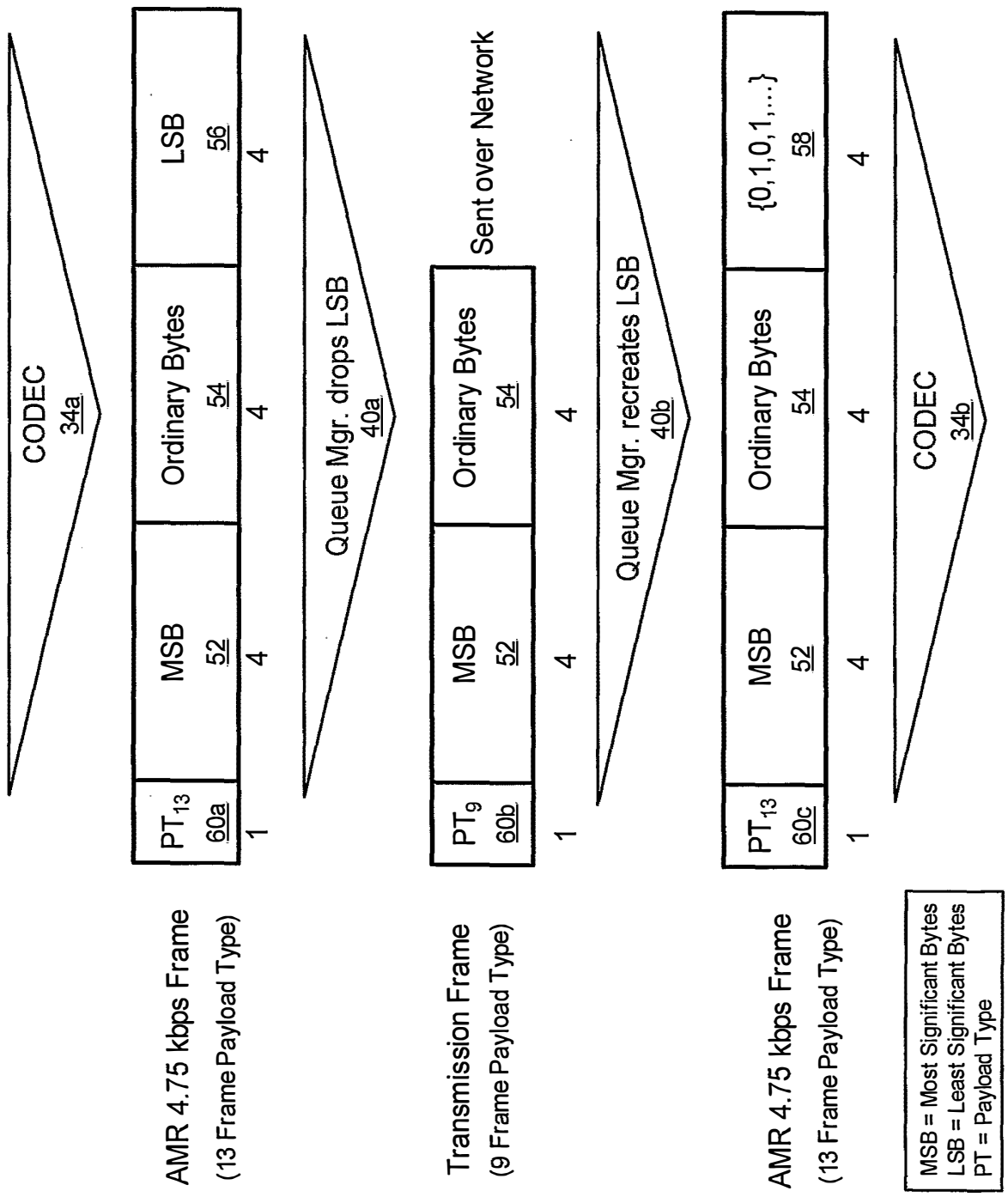

CODEC
34a
AMR 4.75 kbps Frame
(13 Frame Payload Type)
PT13
60a
MSB
52
Ordinary Bytes
54
LSB
56
1
4
4
4
Queue Mgr. drops LSB
40a
Transmission Frame
(9 Frame Payload Type)
PT9
60b
MSB
52
Ordinary Bytes
54
Sent over Network
1
4
4
Queue Mgr. recreates LSB
40b
AMR 4.75 kbps Frame
(13 Frame Payload Type)
PT13
60c
MSB
52
Ordinary Bytes
54
{0,1,0,1,...}
58
1
4
4
4
CODEC
34b
MSB = Most Significant Bytes
LSB = Least Significant Bytes
PT = Payload Type

FIG. 7

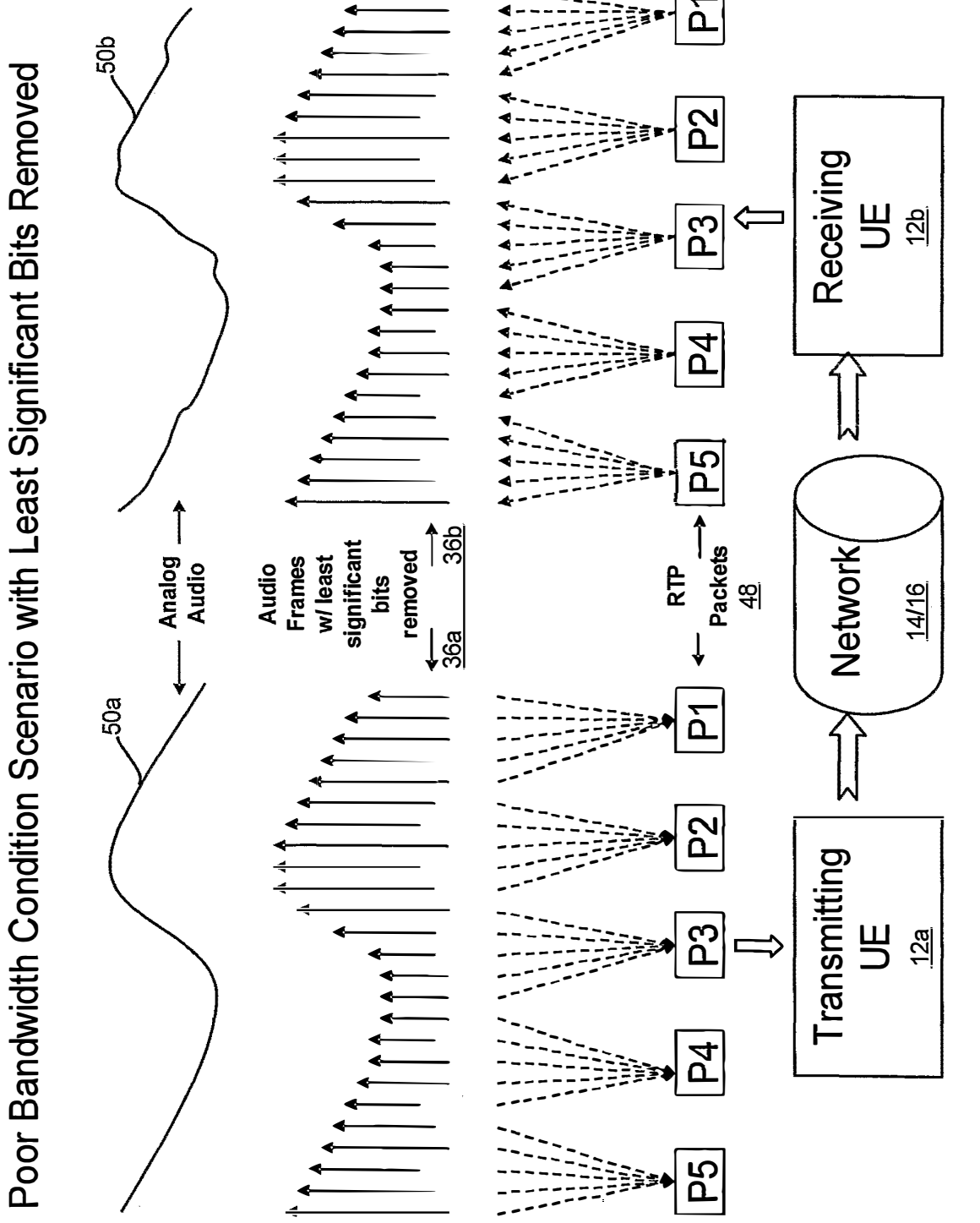

Poor Bandwidth Condition Scenario with Least Significant Bits Removed
50a
50b
Analog Audio
Audio Frames w/ least significant bits removed
36a
36b
RTP Packets
48
P1
P2
P3
P4
P5
Transmitting UE
12a
Network
14/16
Receiving UE
12b

FIG. 8

(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
19 July 2007 (19.07.2007)

PCT

(10) International Publication Number
WO 2007/081929 A2

(51) International Patent Classification:
*H04M 1/64* (2006.01)

(21) International Application Number:
PCT/US2007/000487

(22) International Filing Date: 8 January 2007 (08.01.2007)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
60/756,686 6 January 2006 (06.01.2006) US

(71) Applicant *(for all designated States except US)*: **ORATIVE CORPORATION** [US/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US).

(72) Inventors; and

(75) Inventors/Applicants *(for US only)*: **CHATTERJEE, Saurav** [US/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US). **DREWRY, John** [US/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US). **FULLARTON, Paul** [US/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US). **PERFETTO, Josh** [US/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US). **RANA, Hemendra** [IN/US]; 125 S. Market Street Suite 500, San Jose, CA 95113-2281 (US).

(74) Agent: **COURTNEY, Barbara, B.**; PRACITITIONERS OF COURTNEY STANIFORD & GREGORY LLP, P.o. Box 9686, San Jose, CA 95157 (US).

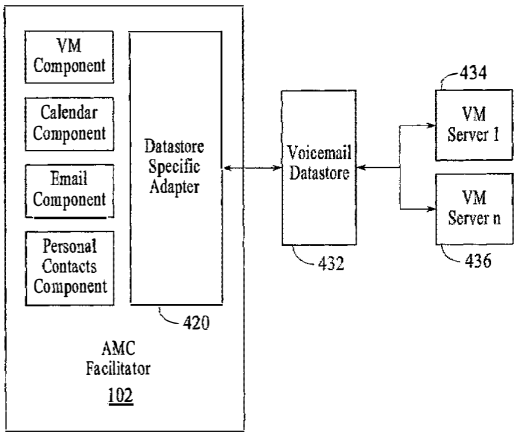
(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LV, LY, MA, MD, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IS, IT, LT, LU, LV, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

**Published:**
— *without international search report and to be republished upon receipt of that report*

*For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.*

(54) **Title:** INTEGRATION OF ENTERPRISE VOICEMAIL IN MOBILE SYSTEMS




400

(57) **Abstract:** An integrated voicemail system is described that provides "visual voicemail" that overcomes the shortcomings associated with a user having to dial into a voicemail system and navigate through a menu of options using conventional voice and/or standard DTMF signaling-based browsing capabilities. The visual voicemail instead provides, via a client device of the user's choice, a visual presentation of a user's voicemails along with state information of the voicemails. The visual presentation of the voicemails includes a user interface that allows a user to control or manipulate voicemails from the client device, where the manipulation includes playing, pausing, rewinding, fast-forwarding, deleting, and changing a state of each voicemail. The visual voicemail also eliminates the need for a user to periodically call-in to the enterprise voicemail system in order to check for voicemail messages as the messages can be automatically delivered to the user's client device.

WO 2007/081929 A2

# INTEGRATION OF ENTERPRISE VOICEMAIL IN MOBILE SYSTEMS

Inventors:

Saurav Chatterjee
John Drewry
Paul Fullarton
Josh Perfetto
Hemendra Rana

## RELATED APPLICATIONS

This application is a continuation-in-part (CIP) application of United States Patent Application number 10/890,922, filed July 14, 2004, which claims the benefit of United States Patent Application number 60/487,143, filed July 14, 2003.

This application claims the benefit of United States Patent Application number 60/756,686, filed January 6, 2006.

This application is related to United States Patent Application number 11/509,109, filed August 23, 2006, which claims the benefit of United States Patent Application numbers 60/710,998, 60/711,051, and 60/711,053, all filed August 23, 2005.

This application is related to United States Patent Application number 11/509,474, filed August 23, 2006, which claims the benefit of United States Patent Application numbers 60/710,998, 60/711,051, and 60/711,053, all filed August 23, 2005.

## TECHNICAL FIELD

The disclosure herein relates generally to communication systems and, in particular, to wireless communication systems.

## BACKGROUND

Mobile communications in today's real-time enterprise can be challenging. The problem is further complicated by changes in the workplace which have led to a more geographically dispersed and highly mobile workforce. In spite of the popularity of electronic mail (email), large numbers of people and employees still depend upon numerous other types of communications to collaborate with colleagues and drive business success. This is especially true for those in sales, service, operations and

management roles who rely upon timely access to and coordination with colleagues as well as other employees, customers, partners and suppliers. Thus, communications remain an essential means of conducting business and staying in contact.

As a result of communications being so critical to business today, many professionals and enterprise employees now handle very large numbers of communications each business day. These communications can include disparate types of communications like emails, voicemails, instant messaging to name a few. Managing these large numbers and disparate types of communications consumes large amounts of time during the typical business day. For the growing number of people who spend a significant part of their day away from their offices or in meetings or other events, managing this large number of communications is highly time-consuming, frustrating and inefficient. Consequently, there is a need for communication systems that provide efficient, timely, and proactive real-time management of multiple types of communications.

## INCORPORATION BY REFERENCE

Each publication, patent, and/or patent application mentioned in this specification is herein incorporated by reference in its entirety to the same extent as if each individual publication and/or patent application was specifically and individually indicated to be incorporated by reference.

## BRIEF DESCRIPTION OF THE DRAWINGS

**Figure 1** is a block diagram of an active mobile collaboration (AMC) system, under an embodiment.

**Figure 2** is a block diagram of a communications system that includes an AMC system, under an alternative embodiment.

**Figure 3** is a block diagram of a communications system that includes an AMC system, under another alternative embodiment.

**Figure 4** is a block diagram showing an integrated system that includes the AMC facilitator integrated with an enterprise voicemail or information system, under an embodiment.

**Figure 5** is a block diagram showing an integrated system that includes the AMC facilitator integrated with an Exchange voicemail database or datastore of the enterprise voicemail system, under an embodiment.

**Figure 6** is a block diagram showing an integrated system that includes the AMC facilitator integrated with multiple enterprise voicemail systems and an Exchange enterprise calendar and contact system, under an embodiment.

**Figure 7** is a block diagram showing an integrated system that includes the AMC facilitator integrated with multiple enterprise voicemail systems and an Exchange enterprise calendar and contact system, under another alternative embodiment.

**Figures 8A, 8B, and 8C** show aspects of the visual voicemail presentation on a client device, under an embodiment.

**Figure 9** is a flow diagram of integrated communications between mobile communication devices and enterprise voicemail systems, under an embodiment.

**Figure 10** is a flow diagram for an Active Call (AC), under an embodiment.

**Figure 11** is a template of the Notification of Active Call Request alert message (AC-1a), under an embodiment.

**Figure 12** is a template of the Notification of Active Call Request Acceptance alert message (AC-1b), under an embodiment.

**Figure 13** is a template of the Active Call Request to a Non-Subscriber (ON-1a), under an embodiment.

**Figure 14** is a message flow diagram for an active call flow, under an embodiment.

**Figure 15** is a block diagram of an AMC system, under an alternative embodiment.

**Figure 16** is a block diagram of an AMC system, under another alternative embodiment.

**Figure 17** is a block diagram of an AMC system, under yet another alternative embodiment.

**Figure 18** is a block diagram of an AMC system in an enterprise domain, under another alternative embodiment.

**Figure 19** is a block diagram of an AMC system in a public domain coupled across components of an enterprise domain, under another alternative embodiment.

**Figure 20** is a block diagram of an AMC system in an enterprise domain, under still another alternative embodiment.

**Figure 21** is a block diagram of an active mobile collaboration (AMC) system, under an embodiment.

DETAILED DESCRIPTION

A communication system is described herein that integrates mobile communication devices with enterprise voicemail systems. The system and corresponding methods described below include receiving information of a voicemail message from a voicemail system. The voicemail system is a server-based system of an enterprise. The voicemail message is received at the voicemail system via conventional methods. The communication system of an embodiment automatically synchronizes a state of the voicemail message between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message. The changes in the voicemail message are made via one or more of the mobile communication device and the access component. In addition to synchronizing state (including deleting voicemail from the external mobile communication device), the embodiment enables the user to respond to a voicemail via a phone call, a text message, a SMS, voice IM, or email.

The integrated voicemail described below provides "visual voicemail" that overcomes the shortcomings associated with a user having to dial into a voicemail system and navigate through a menu of options using conventional voice and/or standard Dual Tone Multi-Frequency (DTMF) signaling-based browsing capabilities. The visual voicemail instead provides, via a client device of the user's choice, a visual presentation of a user's voicemails along with state information of the voicemails. The visual presentation of the voicemails includes a user interface that allows a user to control or manipulate voicemails from the client device, where the manipulation includes playing, pausing, rewinding, fast-forwarding, deleting, and changing a state of each voicemail. The visual voicemail also eliminates the need for a user to periodically call-in to the enterprise voicemail system in order to check for voicemail messages as the messages can be automatically delivered to the user's client device as described below.

In the following description, numerous specific details are introduced to provide a thorough understanding of, and enabling description for, embodiments of the communications systems. One skilled in the relevant art, however, will recognize that these embodiments can be practiced without one or more of the specific details, or with other components, systems, etc. In other instances, well-known structures or operations are not shown, or are not described in detail, to avoid obscuring aspects of the disclosed embodiments.

A communication system is provided herein that uses client-server architectures to improve the efficiency of multiple types of communications. The communication system, referred to herein as the active mobile collaboration (AMC) system, includes a facilitator. The facilitator of an embodiment is an application hosted on one or more servers or other processor-based devices, and communicates a portable or mobile communications device via one or more couplings. The facilitator communicates with the AMC client of a host portable device via a network coupling for example. The facilitator of alternative embodiments can be distributed among one or more portable processor-based devices including the same communication devices as the client application.

The AMC system also includes a client. The client, also referred to as the AMC client, is a component application of a variety of processor-based mobile communication devices and telephones. The components of the AMC system function to improve efficiency of communications by allowing communication device users to increase accessibility of enterprise and personal contact information from mobile phones and other personal digital assistants (PDAs), dynamically manage how and when mobile communications take place, intelligently screen messages, regardless of message type, based on identity of a messaging party, urgency, and subject matter, and determine which contacts in a directory are available to talk and which ones choose not to be disturbed, to name a few.

**Figure 1** is a block diagram of an active mobile collaboration (AMC) system 100, under an embodiment. The AMC system 100 includes any number X(n) of communication devices 101 coupled for communication via one or more facilitators 102 and one or more couplings 104. One or more of the communication devices 101 include an AMC client application. Likewise, the facilitator 102, also referred to herein as the AMC server 102, includes a facilitator application. The AMC client and facilitator function to allow users of the communication devices to dynamically manage how and when mobile calls take place, intelligently screen calls based on caller identity, urgency, and subject matter, determine which contacts in a directory are available to talk and which ones choose not to be disturbed, and increase accessibility of enterprise and personal information (e.g., voicemail, contacts, calendars, etc.) from mobile phones. The AMC system 100 of an embodiment also includes couplings with one or more portals 106 and/or one or more databases 108, but is not so limited.

The communication devices 101 and facilitators 102 described herein are processor-based components running or hosting numerous applications or programs. As such, the communication devices 101 and facilitators 102 can include one or more processors (not shown) coupled among any number/combination of components (not shown) known in the art, for example buses, controllers, memory devices, and data input/output (I/O) devices, in any number of combinations.

The communication devices 101 described herein include processor-based electronic devices, for example, cellular telephones, personal computers, portable computing devices, portable telephones, portable communication devices, subscriber devices or units, PDAs, mobile devices, wireless devices, wireline devices, voice over Internet Protocol (VOIP) devices, private branch exchange (PBX) devices, soft clients, and desktop clients to name a few. The communication devices 101, also referred to as handsets, client devices, mobile devices, mobile communication devices, and portable communication devices, can include all such devices and equivalents, and are not limited to the communication devices described above.

The couplings 104 include wired couplings, wireless couplings, and hybrid wired/wireless couplings, but are not so limited. Furthermore, the couplings 104 can include various networks and/or network components (not shown) of a communication service provider or carrier, but are not so limited. The network and corresponding network components, when present in the couplings 104, can be any of a number of network types known in the art including, but not limited to, local area networks (LANs), metropolitan area networks (MANs), wide area networks (WANs), proprietary networks, backend networks, and the Internet.

**Figure 2** is a block diagram of a communications system 200 that includes an AMC system, under an alternative embodiment. The AMC system includes a facilitator 202 and a client 222 as described elsewhere herein. The facilitator 202 can be one or more facilitators that form a facilitator server cluster 204 and/or database cluster 206 within the enterprise 200E that are resident behind the enterprise firewall 200F, but the AMC system is not so limited. The host enterprise 200E also includes numerous other components, for example, corporate directories and servers 250, authentication servers 252, and enterprise management consoles 254 to name a few. The facilitator 202 is an integrated component of the host enterprise 200E and as such integrates with one or more components of the enterprise 200E. For example, couplings between the facilitator 202 and messaging and collaboration servers (e.g.

Microsoft® Exchange) and/or corporate or other directories of the enterprise 200E allow easy, over-the-air download of personal and corporate contact information to devices, as well as searching of personal and corporate contact directories from the device. Other information of the enterprise 200E can also be delivered to the devices using the AMC system, information including but not limited to calendar information, calendar alerts, calendar reminders, etc.

The facilitator 202 couples to a device of one or more users via one or more network couplings. As an example, the facilitator 202 couples to devices using one or more service provider networks 200S. In this example, the facilitator 202 couples to one or more service provider networks or infrastructures 200S via network couplings 230 (e.g. Internet), and then couples to devices 200M via the respective service provider networks 232. The AMC system protects data transfers between the facilitators 202 and the devices 200M using secure couplings, for example, protected with end-to-end security protocols like Secure Sockets Layer (SSL) or Transport Layer Security TLS cryptographic protocols.

The devices 200M of an embodiment include the AMC client 222. The AMC client 222, also referred to as the client 222, includes a graphical user interface 224 that integrates with the device applications and allows users to receive and scan enterprise information of the enterprise 200E. The enterprise information includes voicemail information (e.g., voicemail messages), contact information, directory information, calendar information, alerts that can include calendar reminders, conference notifications and call requests from colleagues, as described herein and in the Related Applications. Call requests include relevant details such as name, urgency, and subject matter to help users move business forward while screening out unwanted interruptions. The client 222 further provides a presence-aware phonebook that lets users find a contact and determine if the contact is available to talk, even before placing a call. The client 222 eliminates the need to manually enter contacts into the host device 200M. Instead, users download personal and/or corporate contact information over-the-air to their devices. The facilitator 202 and client 222 of the AMC system therefore provide automated, two-way synchronization to ensure contacts are backed up and up to date at the enterprise 200E.

An example of the AMC system of an embodiment is available as the Orative Enterprise Software from Orative Corporation of San Jose, California. The facilitator is available as the Orative Enterprise Server (e.g. runs on a standards-based, Java 2,

Enterprise Edition (J2EE) platform that operates securely behind the enterprise firewall). The client is available as the Orative Client Software (e.g. runs on a variety of popular mobile devices, and leverages the latest application development environments including Symbian OS, Java and BREW to name a few).

While dynamically managing how and when mobile calls take place and intelligently screening calls based on numerous factors described above, the components of the AMC system also improve efficiency of voice communications by increasing accessibility of enterprise and personal contact information from mobile phones. Components of the AMC system of an embodiment support aggregation and management of contact information from various sources including, but not limited to, directories resident on desktop computers, corporate/enterprise directories, and contact information of the mobile device native phonebook, and provide data coupling between those sources and mobile devices hosting the AMC client. This contact information is managed by providing the user with access via the mobile device to dynamically integrated contacts of a contact list and a number of phonebooks from multiple sources. The dynamic integration of multiple disparate directories provided by the AMC system of an embodiment allows a user to indicate the contacts he/she desires among all directories of a corresponding enterprise server, and then dynamically synchronizes all enterprise directories so as to place the desired information from the directories together into a common AMC phonebook, as described in detail below.

**Figure 3** is a block diagram of a communications system 300 that includes an AMC system, under another alternative embodiment. The communications system 300 includes enterprise components, with which the AMC system is integrated, coupled to client devices via a communication or network infrastructure. The enterprise components include, but are not limited to, one or more of a corporate directory, Personal Information Manager (PIM) server, presence server, Private Branch Exchange (PBX) server, management console, and voicemail system (e.g., voicemail servers), to name a few.

The AMC system includes a facilitator as described herein. The facilitator includes an adapter or adapter framework by which the facilitator simultaneously integrates with components of the enterprise and enterprise servers. The facilitator uses an adapter for each enterprise component to which it integrates. The adapter of an embodiment is a protocol-specific adapter for each enterprise component to which it integrates; alternatively, the adapter includes vendor-specific adapters. The facilitator

integrates with multiple components simultaneously. The AMC adapters convert the data from the enterprise components (e.g. external) into a common data structure.

The facilitator includes one or more applications that support multiple functions provided by the AMC system. The AMC system functions include integrated access to enterprise voicemail as described here and, additionally, text messaging, pre-call management, appointments and contacts, notifications, availability (presence), voicemail, and PBX remote control as described in the Related Applications.

The facilitator couples to a client device of one or more users via one or more network couplings or infrastructures. As an example, the facilitator couples to a mobile network using a coupling with another communications network (e.g. Internet). The mobile network or mobile infrastructure, which includes one or more service provider networks associated with respective ones of the client devices, provides a coupling to individual client devices.

Communications between the facilitator and the client device are controlled by the facilitator using one or more components and applications. The functions provided by the facilitator in controlling communications include one or more of rate control, synchronization (sync), call signaling, data transfer, over the air (OTA) provisioning, and device management to name a few. Optionally, the communications path between the facilitator and the communications network includes an AMC proxy server.

The AMC system of an embodiment described above extends the reach of the corporate voicemail to portable electronic devices through integration of the client device with the enterprise voicemail systems. The AMC system thus provides access via a user's client device to some or all of the features available from the user's desk telephone or enterprise telephone. The user's client device includes but is not limited to client devices operating inside the enterprise (e.g., Internet Protocol (IP) telephones, WiFi devices, etc.) as well as client devices operating external to the enterprise (e.g., cellular telephones, wireless devices, personal digital assistants, Worldwide Interoperability for Microwave Access (WiMAX) devices, WiFi devices, etc.). The enterprise voicemail system including the user's desk telephone or enterprise telephone includes but is not limited to PBX systems and central exchange (Centrex) systems. The integration of the AMC system with the enterprise voicemail systems is a two-way integration in which updates to voicemail message box states and message states at the voicemail server (e.g., enterprise) are reflected on the client device, and updates to voicemail message box states and message states made at the client device are reflected

on the enterprise voicemail server. The features of the enterprise telephone system available via the client device are described below first in the context of usage patterns.

A usage example under an embodiment includes a user receiving a voicemail at the enterprise voicemail system. Information of the voicemail message is gathered by the facilitator, which in turn propagates it to the AMC client which, in response to the voicemail, displays or presents a view with the voicemail notification on the client device. The voicemail notification includes identification information of the originator of the voicemail message, including phone number and name. The AMC client presents the user with both response options and playback options, depending on the caller identification and/or AMC subscriber status of the caller leaving the "new" voicemail message. The response options include responding to the voicemail originator via telephone call, email message, instant message, and/or voice instant message or forwarding the voicemail. The voicemail playback options include manipulation of the voicemail message via the user interface of the client device, the manipulations including playing, pausing, rewinding, and fast-forwarding the voicemail message as described in detail below. Because the user is notified via the client device of voicemail messages waiting in the enterprise voicemail system, the user is not required to dial-in to the enterprise system to access voicemail information.

Another usage example under an embodiment includes a user playing a voicemail message via the AMC client of the client device. As in conventional voicemail systems, the user can dial into the voicemail system from the client device and use standard DTMF signaling-based browsing capabilities to manipulate (e.g., play) all voicemail. However, the AMC system alternatively allows the user to play a specific voicemail message using the AMC client. When playing voicemail with the AMC system, the AMC client downloads an audio voicemail file from the AMC facilitator and plays the downloaded voicemail at the client device. The facilitator of an embodiment converts the content type of the voicemail message to an audio format best for the client device before or simultaneous with file transmission to the client. Playback of the voicemail message also includes options to pause, rewind, and fast-forward the voicemail during playback. The AMC system changes the voicemail notification state from "unread" to "read" in response to the downloading and/or playing from the client device and propagates the changed state information to the enterprise voicemail system.

Another usage example of an embodiment includes use of the AMC application

to respond to a voicemail message. When responding to a voicemail message, the AMC system allows a user to generate and send from the user's client device a call request or a text message to the caller. Furthermore, if the caller's telephone number is known, the user may dial the caller.

The AMC system also allows a user to delete voicemail messages from the enterprise telephone system using the AMC client. In response to user action to delete a voicemail message via the AMC client, the voicemail message is deleted from the enterprise voicemail system and deleted from the AMC system.

If not deleted, voicemail messages eventually expire in the AMC system as described herein. Upon expiration of a voicemail message in the AMC system, the voicemail appears on the client device via the AMC client as an expired conversation, but remains available to be played. After a period of time, the expired conversation cleanup of the AMC system deletes the voicemail notification from the AMC client. In one embodiment, the message continues to be available in the voicemail server so that the enterprise telephone system can allow the user to play the voicemail; in another, the VM is moved to an archive location, but deleted from the VM system.

The voicemail integration of the AMC system includes integration of the enterprise voicemail server to the AMC facilitator and integration of the user voicemail interaction with AMC client interaction. This integration is described in detail below.

Integration of the enterprise voicemail server to the AMC facilitator includes synchronization between the enterprise voicemail server and the AMC system such that any changes to voicemails introduced via any interface to the voicemail system are propagated to any other interface. Thus, for example, changes introduced to a voicemail via a desk telephone are propagated to one or more of a unified messaging client and the AMC system (e.g., client device). Similarly, changes introduced to a voicemail via a unified messaging client are propagated to one or more of the desk telephone and the AMC system (e.g., client device). Likewise, changes introduced to a voicemail via the AMC system (e.g., client device) are propagated to one or more of the desk telephone and the unified messaging client.

**Figure 4** is a block diagram showing an integrated system 400 that includes the AMC facilitator 102 integrated with an enterprise voicemail system 420, under an embodiment. The facilitator 102 of an embodiment interfaces or integrates with components of enterprise voicemail systems using a datastore adapter 420 that is coupled to the voicemail datastore 432 of the enterprise. The enterprise voicemail

datastore 432 is coupled to one or more enterprise voicemail servers 434-436 as appropriate to the host enterprise configuration.

The datastore adapter 420 leverages the AMC adapter and component architecture described herein. The datastore adapter 420 is an adapter module or component that is coupled to and/or a component of the facilitator 102 and is configured specifically for a type or configuration of the voicemail datastore 432 of the enterprise. The datastore adapter 420 is coupled to one or more components or modules of the AMC system including but not limited to a voicemail (VM) component, a calendar component, an electronic mail (email) component, and a personal contacts component to name a few. The VM component includes components or code specific to one or more of a type or configuration of the enterprise voicemail systems and is independent of the enterprise voicemail servers 434-436. The VM component interfaces with the enterprise voicemail servers 434-436 using the datastore adapter 420.

Many enterprise voicemail systems use Microsoft Exchange as a voicemail datastore. **Figure 5** is a block diagram showing an integrated system 500 that includes the AMC facilitator 102 integrated with a Microsoft Exchange ("Exchange") voicemail datastore 532 of the enterprise voicemail system, under an embodiment. The Exchange datastore 532 typically makes the enterprise voicemail available to the AMC facilitator 102 via the Inbox folder and sub-folders of Exchange.

The facilitator 102 of the system 500 interfaces or integrates with enterprise voicemail systems using an Exchange adapter 520 that is coupled to the Exchange application 532 of the enterprise. The enterprise Exchange 532 is coupled to one or more enterprise voicemail servers (e.g., Cisco Unity server 534, Avaya Unified messenger server 536, etc.) as appropriate to the host enterprise configuration. The Exchange adapter 520 leverages the AMC adapter and component architecture described herein. The Exchange adapter 520 is an adapter module or component that is coupled to and/or a component of the AMC facilitator 102 and is configured specifically for the enterprise Exchange 532. The Exchange adapter 520 is coupled to one or more components or modules of the AMC system including but not limited to a voicemail (VM) component, a calendar component, an electronic mail (email) component, and a personal contacts components to name a few. The VM component includes components or code specific to Exchange 532 and is independent of the enterprise voicemail servers 534-536. The VM component interfaces with the

enterprise voicemail servers 534-536 using the Exchange adapter 520.

The voicemail of the system 500 is available to the AMC facilitator 102 via the Inbox folder and sub-folders as described above. Any updates to the voicemail message state (e.g., read, unread, erased, etc.) are immediately reflected in the enterprise Exchange 532. Therefore, by synchronizing the AMC system with Exchange 532, both the contents of the voicemail (e.g., sender, time, voice message, etc.) and its state are available to the AMC system. In addition to providing the AMC system all of the information of interest, this configuration allows the AMC system to integrate and function with many voicemail systems, since these server-centric voicemail systems provide integration to Exchange 532. Both Cisco Unity 534 and Avaya Unified Messenger/Modular Messaging 536 support this configuration, to name a few.

Further, some enterprise configurations may not include unified messaging and therefore may use a different server for each of their voicemail and email datastores. Consequently, the AMC system of an embodiment is configured to communicate with at least one enterprise server to fetch voicemails and with at least one enterprise server to fetch contacts and appointments, as described below.

Enterprise voicemail systems can include configurations using numerous different voicemail systems and using Exchange as a calendar and contact datastore. **Figure 6** is a block diagram showing an integrated system 600 that includes the AMC facilitator 102 integrated with multiple enterprise voicemail systems 634-638 and an Exchange enterprise calendar and contact system 632-E, under an embodiment.

The facilitator 102 of an embodiment interfaces or integrates with enterprise voicemail systems using a dedicated datastore adapter that is coupled to the voicemail datastore of each voicemail system of the enterprise. At the enterprise, each enterprise voicemail datastore is coupled to an associated voicemail server or system as appropriate to the host enterprise configuration. At the AMC facilitator 102 , the VM component of the facilitator is coupled to each of first 620-1, second 620-m, and third 620-n datastore adapters. The first datastore adapter 620-1 is coupled to first 634-1 and second 634-2 enterprise voicemail servers or systems via a first enterprise voicemail datastore 632-1. The second datastore adapter 620-m of the facilitator 102 is coupled to a third enterprise voicemail server or system 636 via a second enterprise voicemail datastore 632-n. Similarly, the third datastore adapter 620-n of the facilitator 102 is coupled to a fourth enterprise voicemail server or system 638 via a third enterprise

voicemail datastore 632-m.

Each of the first 620-1, second 620-m, and third 620-n datastore adapters (collectively 620) of the facilitator 102 leverages the AMC adapter and component architecture described herein. The datastore adapters 620 are adapter modules and/or components that are coupled to a component of the facilitator 102 and are configured specifically for a type or configuration of the respective first, second, and third voicemail datastores (collectively 632) of the enterprise. The VM component includes components or code specific to the four enterprise voicemail systems and is independent of the enterprise voicemail servers 634-638. The VM component interfaces with the four enterprise voicemail servers 634-638 using the three datastore adapters 620-1/620-m/620-n.

Regarding calendar and contacts information, the facilitator 102 of the system 600 interfaces or integrates with enterprise contact and calendar systems using an Exchange adapter 620-E that is coupled to the Exchange application 632-E of the enterprise, but is not so limited. The enterprise Exchange 632-E is coupled to one or more enterprise components as appropriate to the host enterprise configuration. The Exchange adapter 620-E leverages the AMC adapter and component architecture described herein. The Exchange adapter 620-E is an adapter module or component that is coupled to and/or a component of the AMC facilitator 102 and is configured specifically for the enterprise Exchange 632-E. The Exchange adapter 620-E is coupled to a calendar and contacts component of the facilitator 102. The calendar and contacts component includes components or code specific to Exchange 632-E and is independent of the enterprise calendar and contacts servers. The calendar and contacts component interfaces with the enterprise calendar and contacts servers using the Exchange adapter 620-E.

Some enterprise configurations use a different server type for each of their voicemail and email datastores, and the AMC system of an embodiment is configurable to communicate with at least one enterprise server to fetch voicemails and with at least one enterprise server to fetch contacts and appointments, as described above. Further, some voicemail systems (e.g., Octel) use an embedded datastore in the voicemail system instead, and the AMC system includes a configuration having a proprietary adapter that is voicemail server-specific (e.g., Octel server-specific). Additionally, some voicemail systems (e.g., Avaya) do not integrate to Exchange or Domino but instead use a proprietary datastore and an Internet Message Access Protocol (IMAP)

version 4 (IMAP4) interface, and the AMC system includes a configuration that supports IMAP directly from the facilitator. These alternative configurations are described below.

**Figure 7** is a block diagram showing an integrated system 700 that includes the AMC facilitator 102 integrated with multiple enterprise voicemail systems and an Exchange enterprise calendar and contact system, under another alternative embodiment. The facilitator 102 of system 700 generally interfaces or integrates with enterprise voicemail systems using a dedicated datastore adapter that is coupled to one of a voicemail datastore or a voicemail system of the enterprise as appropriate to a system configuration of the enterprise. When a voicemail system uses a voicemail datastore, the voicemail datastore is coupled to an associated enterprise voicemail server or system.

The enterprise of integrated system 700 includes numerous different voicemail systems, including Cisco Unity, Avaya Unified Messenger, Octel, and Avaya Intuity systems. Each of the Cisco Unity and Avaya Unified Messenger systems use an Exchange datastore 732-1, and the facilitator 102 interfaces or integrates with enterprise voicemail systems using an Exchange adapter 720-1 that is coupled to the Exchange 732-1 of the enterprise. The enterprise Exchange 732-1 is coupled to each of the Cisco Unity 734-C and Avaya Unified 734-AU messenger servers. The Exchange adapter 720-1 leverages the AMC adapter and component architecture described herein. The Exchange adapter 720-1 is an adapter module or component that is coupled to and/or a component of the AMC facilitator 102 and is configured specifically for the enterprise Exchange 732-1. The VM component includes components or code specific to Exchange 732-1 and is independent of the enterprise voicemail servers 734-C/734-AU. The VM component interfaces with the enterprise voicemail servers 734-C/734-AU using the Exchange adapter 720-1.

The voicemail of the enterprise is available to the AMC facilitator 102 via the Inbox folder and sub-folders as described above. Any updates to the voicemail message state (e.g., read, unread, erased, etc.) are immediately reflected in the enterprise Exchange 732-1. Therefore, by synchronizing the AMC system with Exchange 732-1, both the contents of the voicemail (e.g., sender, time, voice message, etc.) and its state are available to the AMC system. In addition to providing the AMC system all of the information of interest, this configuration allows the AMC system to integrate and function with the Cisco Unity 734-C and Avaya Unified

Messenger/Modular Messaging 734-AU systems.

The Avaya Intuity voicemail system uses a proprietary datastore and an IMAP4 interface. The facilitator 102 interfaces or integrates with the Avaya Intuity enterprise voicemail system using an IMAP4 adapter 720-I that supports IMAP directly from the facilitator 102. The IMAP4 adapter 720-I is coupled to the VM component of the facilitator 102 and is coupled to the Avaya datastore 732-A of the enterprise. The Avaya datastore 732-A is coupled to the Avaya Intuity voicemail server 734-AI as appropriate to the host enterprise configuration.

The Octel voicemail system uses an embedded datastore. The facilitator 102 interfaces or integrates with the Octel voicemail system using an proprietary adapter 720-P that supports the Octel system directly from the facilitator 102. The proprietary adapter 720-P is coupled to the VM component of the facilitator 102 and is coupled to the Octel voicemail server 734-O of the host enterprise. Similar interfaces can be provided under the disclosure herein for voicemail systems and server other than those described above.

Regarding calendar and contacts information, the facilitator 102 of the system 700 interfaces or integrates with enterprise contact and calendar systems using an Exchange adapter 720-E that is coupled to the Exchange 732-E of the enterprise. The enterprise Exchange 732-E is coupled to one or more enterprise components as appropriate to the host enterprise configuration. The Exchange adapter 720-E leverages the AMC adapter and component architecture described herein. The Exchange adapter 720-E is an adapter module or component that is coupled to and/or a component of the AMC facilitator 102 and is configured specifically for the enterprise Exchange 732-E. The Exchange adapter 720-E is coupled to a calendar and contacts component of the facilitator 102. The calendar and contacts component includes components or code specific to Exchange 732-E and is independent of the enterprise calendar and contacts servers. The calendar and contacts component interfaces with the enterprise calendar and contacts servers using the Exchange adapter 720-E.

The facilitator of the integrated systems 400-700 described above generally uses a protocol specific to the datastore to read, update, or delete voicemails. If the datastore is Exchange, then the facilitator can use either WebDAV, MAPI, or IMAP for example. The facilitator can either periodically poll the appropriate folder or subfolders in Exchange using one of these protocols or it may configure Exchange to send it a signal, using WebDAV or IMAP, when a voicemail arrives, or one is deleted, or when a

voicemail's state changes; once the facilitator receives the signal from Exchange, the facilitator retrieves the new information.

If the datastore is not Exchange but does supports IMAP, then the interaction is the same as above, but using IMAP only instead of WebDAV and/or MAPI.

If the datastore uses a proprietary protocol, then the facilitator periodically polls the appropriate data structure hierarchy using this protocol. If a push mechanism is supported, the datastore sends a signal to the facilitator, using WebDAV or IMAP, when a voicemail arrives, or one is deleted, or when a voicemail's state changes; once the facilitator receives this signal, the facilitator retrieves the new information.

As described above, the AMC system provides notifications to a client device of a voicemail message received at the enterprise. In providing notifications of the voicemail, the AMC system of an embodiment provides synchronization (e.g. one-way and/or two-way) between the enterprise voicemail server and the AMC facilitator. In so doing, the AMC facilitator periodically fetches voicemail information from the enterprise voicemail server. In addition to fetching voicemail information, the facilitator of an embodiment can receive notifications when there is a change in state in the enterprise voicemail server. Using the received voicemail information, the AMC system provides voicemail notifications to a user that has received voicemails by either starting a new conversation thread in the AMC client when a caller sends a voicemail to the user, or, inserting the voicemail into a previous conversation if the caller has sent previous voicemails or other messages or calls. The voicemail may automatically expire after a configurable time (e.g., 24 hours) but is not so limited.

The starting of a new conversation thread, or insertion into an existing one, includes transmission by the facilitator of a Orative Markup Language (OML) voicemail message to the device, over the OLWP protocol, as described in the Related Applications, but is not so limited. The message includes information of the voicemail message received in the enterprise voicemail system but is not so limited. The underlying mechanism used by OLWP for signaling a client of new information includes the use of a persistent TCP connection or a standard push message including, but not limited to, an SMS message, email message, MMS, or any other push message type accepted by a particular target device; the client then uses OLWP and OML to fetch the new voicemail information from the facilitator.

Before transmitting to the client, using information of the received voicemail, the facilitator maps the voicemail originator's (caller) automatic number identification

(ANI) or email to either an AMC user handle or to a name in the user's contact directories. The mapping uses contact information of one or more of a variety of available sources, including the corporate directories, the user's personal contacts, and potential CRM databases.

If the caller ANI or email address match that of an AMC user, the facilitator determines the caller is an AMC user and sends one or more of the originator's AMC address, active AMC mobile telephone number, and official first and last names to the client in a voicemail message. If the originator is not an AMC user, but the email address or caller ID matches a contact in the user's corporate phonebook (e.g., the full Active Directory), the originator's name, email address, and one or more telephone fields is retrieved and sent as part of a voicemail message to the client. The specific telephone fields retrieved are configurable; the default setting is the corporate desk telephone. If the originator is not an AMC user, but the caller ID matches a contact in the user's personal contact directory (not just the active telephone book), the originator's name, email address, and additional phone numbers are retrieved and sent as part of the voicemail message to the client. If the caller ID is unknown because the information does not match any contact information related to the user in the AMC system, an "unknown caller" message is sent from the facilitator to the client along with the number (if known). The importance of the voicemail is also sent by the facilitator to the client.

The AMC system supports several types of user and response options upon receiving at the client device notification of an enterprise voicemail, and the response options of an embodiment depend on the type of caller and the desires of the user recipient of the message. These options include an enterprise voicemail box dial-in option, a voicemail playback option, and reply options by which the user generally can reply to a voicemail using an active call request, text message, or call options as described below. The options also include voicemail message delete options.

For example, if the originator leaving the voicemail is an AMC user, the client can respond with a call request or text message, and the facilitator includes the originator's AMC address in the call request. If the telephone number is known for the originator of the voicemail, the client can initiate a call to the caller by dialing the caller's telephone number. If the telephone number is known but the originator is not an AMC subscriber, the client can respond with a call request or text message via a communication path to non-subscribers as described in the Related Applications.

The AMC system supports a voicemail playback option under which the AMC client dials into the enterprise voicemail system and leverages the existing voice mail server to navigate among the user's voicemails. The user enters his/her enterprise voicemail ID and password, and is then enabled through the use of DTMF signaling (e.g., using client device key entries) to perform actions including navigating to a selected message, playing the message, pausing the message, rewinding the message to name a few. The voicemail telephone number is set on the user portal as a preference.

The AMC system of an embodiment provides several options for play back of voicemail messages. A first playback option, for example, includes the AMC client commanding the facilitator to couple or link a telephone number (e.g., enterprise telephone number) to the voicemail message, and the client dials into the telephone number to play the message. A second playback option includes the AMC client commanding the facilitator to subsequently dial back the AMC client and play the voicemail message. The first and second playback options include the use of a direct integration to a private branch exchange (PBX), a third-party VoiceXML (VXML) server, and/or other communication system.

A third playback option includes the facilitator transmitting the voicemail message directly to the AMC client; the client stores the received message and plays it as described herein. The third playback option includes downloading via streaming, bulk downloading of voicemail messages, and/or downloading byte ranges of voicemail data; other transmission alternatives can be used as known in the art. Downloading of voicemails to the client can occur automatically and/or in response to a request or other user action. In one embodiment, the client may download a certain category of voicemails automatically (e.g., from certain people or high priority ones) before presenting the voicemail to the user, therefore eliminating or at least substantially reducing network latency before the user can play the message. In other cases, the client may only download each voicemail only after the user explicitly asks to play the voicemail.

Furthermore, voicemails can be downloaded under a variety of circumstances including but not limited to downloading prior to the client sending notification of the voicemail to a user, downloading simultaneous with sending notification of the voicemail to a user, and/or downloading after sending notification of the voicemail to the user. One or more voicemails can be downloaded to the client in a download session, and one or multiple voicemails can be stored simultaneously on the device.

The various alternatives for downloading and/or storing voicemail messages can be configurable or reconfigurable by a user of the client device and/or automatically configurable or re-configurable by the facilitator in response to one or more parameters (e.g., data tariff charges, importance of a voicemail, voicemail sender, etc.). The third playback option may be preferred from the user experience perspective, as it operates independently of PBX vendors, and provides a long-term path that enables the addition of more features (e.g., voice subjects) and optimizes the user experience.

For playing back a specific voicemail message under the third playback option, the client and facilitator, in an embodiment, supports a synchronous request-response file download mechanism. Under the file download mechanism, the facilitator embeds the reference ID and byte length of a binary file in a voicemail-type call request in the original OML message, where the binary file includes the voicemail contents. The AMC client then requests the binary file via HTTP over OLWP to the facilitator; the client may optionally specify a byte offset and length of the binary file. The facilitator responds to the client file request with a HTTP over OLWP response comprising the requested byte range for the file. HTTP over OLWP is identical to standard HTTP over TCP, except over a multiplexing OLWP protocol, which in turn is over TCP; the benefit of OLWP is that it allows the client to maintain a single TCP connection to the facilitator, over which it transmits both OML and HTTP data.

If the connection between the client and the facilitator is interrupted during downloading of the file, the client re-requests the appropriate file or byte range within the file. In an embodiment, the client repeats the process described above to download the file, but is not so limited.

Upon completing download of the binary file with the voicemail message content, the AMC client stores the voicemail message in the client device. This allows a user to access the voicemail message and perform actions like playing, pausing, rewinding, and re-playing the message without any added network latency. In one embodiment, the binary file that includes the message is over-written when a new binary file including a new voicemail message is downloaded to the client. To re-access a voicemail message that has been over-written, the client can request and fetch the corresponding binary file again from the facilitator. In another embodiment, multiple binary files are stored or cached on the client, so all of these voicemails can be played without added network latency; standard caching algorithms (e.g., least recently used) are used to prioritize which files are overwritten first when memory constraints

arise.

The AMC system of an embodiment is configured to convert the voicemail message among any number of formats including Waveform audio format (WAV), Code Excited Linear Prediction (CELP), Adaptive Multi-Rate (AMR), and Audio Interchange File Format (AIFF) to name a few; alternative embodiments may support other audio formats. The facilitator of an embodiment converts the content type of the voicemail message to an audio format more appropriate for the client device and/or a communication network by which the client device communicates before or simultaneous with file transmission to the client. The determination as to whether to convert an audio format is made according to one or more parameters of the client device and/or the communication network. For example, the decision on transcoding of the audio format is made using information of one or more of audio file size, client device capabilities, communication network bandwidth, communication network capabilities, and communication network data tariff charges.

The AMC system of an embodiment is configured to allow a user to delete a voicemail message in the enterprise voicemail system by deleting the message on the client device. In deleting a message, the AMC client locally deletes the voicemail notification. Embodiments of the AMC system provide two-way synchronization so that a delete action taken by a user on the AMC client automatically results in the facilitator deleting the voicemail at the appropriate voicemail server in the enterprise voicemail system. In an alternative embodiment a delete action taken by a user on the client device may not be synchronized with the facilitator, and thus may not be synchronized with the enterprise voicemail system. If the voicemail server deletes a voicemail (e.g., because the user deleted it using his desk telephone), the facilitator receives an event from the enterprise voicemail system, and deletes the voicemail from the client in response to the event.

The AMC system also provides a status of voicemail messages of the enterprise voicemail system. Generally, the AMC client is configured to change the status of an enterprise voicemail message to "read" or "unread" from the client device. A status change made to an enterprise voicemail message at the client is propagated or communicated to the facilitator which then changes the voicemail status at the enterprise voicemail server to be consistent with the status in the AMC client.

For example, when a user listens (even partially) to an enterprise voicemail from the client device, the status of the enterprise voicemail is automatically changed at

the client to a "read" status; the facilitator then communicates the status change to the enterprise voicemail server where it is also changed to a "read" status.

The integrated voicemail described herein provides "visual voicemail" that overcomes the shortcomings associated with a user having to dial into a voicemail system and navigate through a menu of options using conventional voice and/or DTMF-based browsing. The visual voicemail instead provides a visual presentation on a client device of the user's voicemails along with state information of the voicemails. The visual presentation of the voicemails includes a user interface that allows a user to control or manipulate voicemails from the client device, where the manipulation includes playing, pausing, rewinding, fast-forwarding, deleting, and changing a state of each voicemail.

**Figures 8A, 8B, and 8C** show aspects of the visual voicemail presentation on a client device, under an embodiment. **Figure 8A** shows an example download screen of the client device during downloading of a voicemail to the client device, under an embodiment. **Figure 8B** shows an example playback screen of the client device during playback of a voicemail at the client device, under an embodiment. The playback screen displays information including, for example, the voicemail originator's name, a time the voicemail was sent, a length of the voicemail, and the progress of the playback. **Figure 8C** shows controls displayed on the client device and the corresponding functions for use in manipulating a downloaded voicemail message at the client device, under an embodiment.

**Figure 9** is a flow diagram of integrated communications between mobile communication devices and enterprise voicemail systems, under an embodiment. Information of a voicemail message is received 902 from a voicemail system. The voicemail system is a server-based system of an enterprise. The voicemail message is received at the voicemail system via conventional methods. A state of the voicemail message is automatically synchronized 904 between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message. The changes in the voicemail message are made via one or more of the mobile communication device and the access component.

The AMC system as described above allows a user of the client device (referred to as the "originator" of the Active Call Request in the following description) to use an Active Call Request to politely ask the originator of a voicemail message (referred to as the "recipient" of the call request in the following description) if the recipient is ready

to take a phone call in response to his/her voicemail message. The Active Call Request, which is described in detail in the Related Applications, also provides discreet response options by which the recipient can provide timely feedback to the originator. Callers thus have the satisfaction of knowing the receiver acknowledged their call request and will make time to talk.

The AMC system, upon initiation of an Active Call Request, provides the originator various options to tailor the request to his/her unique situation and to provide more context to the intended recipient(s). Active Call Requests support numerous input options including priority, text subject, voice subject, originator's phone number, and expiration time to name a few. The text subject input option allows end-user keying of a text subject tag. An embodiment supports selection of pre-defined text subjects, and the portal supports definition of the pre-defined text subjects.

The voice subject or voice instant message (VIM) input option supports the user recording an audio voice subject of a pre-specified duration via a handset interface. The voice subjects are stored on the handset to facilitate offline usage, but are not so limited. The voice subject of an embodiment is kept on the originating handset until successful delivery of the voice subject to the facilitator. The facilitator may convert the audio format between the originating handset and the recipient handset, based on individual handset audio format playback capabilities.

The originator's phone number input option supports end-user keying of phone numbers and embedding of these phone numbers in the request alert. The default number is the originator's mobile phone number.

The expiration time input option allows the end-user to input an expiration time value at the handset to specify a time period after which the call request automatically expires. A maximum expiration period is no more than 72 hours in the future, and a default expiration time is 180 minutes, but the embodiment is not so limited.

**Figure 10** is a flow diagram for an Active Call (AC), under an embodiment. An Active Call begins when an originator selects a contact and initiates an Active Call request to the contact using the originator's handset. The facilitator processes the call request and in response forwards an alert message to a handset of the selected contact, where the alert message is referred to herein as a Notification of Active Call Request alert message (AC-1a).

The flow diagrams provided herein include logic flows that start in the upper left-most cell and proceed to the right, then down, and back to the left, row-by-row (see

arrows) until the lower left cell is reached. Arrows are used in the call flow to direct the flow for the reader. Generally, caller/originator behavior is always on the left, AMC facilitator behavior in the middle, and call recipient behavior on the right. Within the call flow the action taken by a party (originator and/or recipient) is shown followed by the possible responses (bulleted). The normal response course is designated with a solid bullet, and alternative courses are designated with hollow bullets.

**Figure 11** is a template of the Notification of Active Call Request alert message (AC-1a), under an embodiment. This alert message is sent to the recipients of an Active Call Request to inform the intended recipient that a caller wishes to speak with him, and provides convenient options for responding to the request. The Notification of Active Call Request alert message supports a number of response options by the recipient including accepting the request with the recipient's handset phone number, accepting the request with a phone number input by the recipient, initiating the mobile call to the originator using the value in the originator's phone number field of the alert message, responding with a pre-defined text message, initiating a voice instant message to the originator, deleting the request, and requesting a postponement of the call. Actions by the recipient including accepting the call request, initiating the mobile phone call, responding with a pre-defined text message, and initiating a voice instant message to the originator result in automatic deletion of the alert message from the recipient's inbox, but the embodiment is not so limited. Further, information of delivery confirmation states of the Notification of Active Call Request alert message is provided to the originator.

The facilitator processes any response by the receiver and in response forwards an alert message to the originator, where the alert message is referred to herein as a Notification of Active Call Request Acceptance alert message (AC-1b). **Figure 12** is a template of the Notification of Active Call Request Acceptance alert message (AC-1b), under an embodiment. This alert message notifies the originator that the intended recipient of the Active Call Request has accepted the request and that the recipient is ready to participate in the call. Further, the Notification of Active Call Request Acceptance alert message provides the originator with response options that include a one-button mechanism for initiating a mobile call to the recipient, and an option to delete the alert message. Additionally, information of delivery confirmation states of the Notification of Active Call Request Acceptance alert message is provided by the facilitator to the receiver.

The AMC system described herein also supports Active Call Requests by an originator/subscriber receiving a voicemail message from a recipient/non-subscriber. The AMC system upon initiation of an Active Call Request by an originator (subscriber) to a recipient (non-subscriber) provides the originator various options to tailor the request to his/her unique situation and to provide more context to the recipient (non-subscriber). Active Call Requests to non-subscribers support numerous input options including priority, text subject, originator's phone number, and expiration time to name a few.

The text subject input option allows the originator (subscriber) to key a text subject tag. An embodiment supports selection of pre-defined text subjects, and the portal supports definition of the pre-defined text subjects.

The originator's (subscriber) phone number input option supports originator (subscriber) keying of phone numbers and embedding of these phone numbers in the request alert. The default number is the originator's (subscriber) mobile phone number. The expiration time input option allows the originator (subscriber) to input an expiration time value at the handset to specify a time period after which the Active Call Request automatically expires.

Once generated, the active call request to the recipient (non-subscriber) is transferred to the originator's (subscriber) handset by the AMC system. The AMC system repackages contents of the Active Call Request into an SMS message directed at the mobile phone number of the recipient (non-subscriber). As one example, the SMS message is as follows: "<Originator name> at <phone number> is requesting to speak with you before <local expiration time>. <Priority>". Delivery confirmation states of the call request SMS message are provided to the originator (subscriber), and these states include at least one of "Not Sent", "In Transit", "Delivered", and "Delivery Failed". **Figure 13** is a template of the Active Call Request to a Non-Subscriber (ON-1a), under an embodiment.

Considering the AMC system protocol, the Active Call flow described above with reference to **Figure 10** is revisited. **Figure 14** is a message flow diagram for an active call flow, under an embodiment. The active call flow includes three types of messages: Active Call Request (ACRequest); Active Call Response (ACResponse); and Delivery Confirmation (DeliveryConfirmation). An active call is initiated with an ACRequest; logically, it is sent by the Sender Client (SC) to the Recipient Client (RC); the SC and RC are both AMC clients hosted on separate handsets. When the RC

receives the ACRequest, it sends a Delivery Confirmation back to the SC. The RC also may send an ACResponse, indicating acceptance or postponement of the corresponding ACRequest. When the SC receives the ACResponse, it also sends back a DeliveryConfirmation to the RC.

The SC and the RC of an embodiment do not communicate directly with each other but rather convey information via one or more facilitators hosted on one or more servers, where the servers are referred to as the Sender Server (SS) and the Recipient Server (RS). The SC connects to the SS while the RC connects to the RS. The SS and RS may be in the same domain or different domains.

To optimize client traffic and reduce bandwidth, Delivery Confirmations are not generated by the AMC clients themselves. Instead, the RS sends a Delivery Confirmation message when it receives the ACK from the AMC client.

Delivery Confirmations are automatically generated by the system, enabling the sender to know of the status of the message. Unlike a Delivery Confirmation, an Active Call Response message is only generated after user input, indicating a response to the active call request. The three components of active calls, Active Call Requests, Active Call Response, and Delivery Confirmations are each described in turn below.

Beginning with the Active Call Request, the SC generates a new Active Call Request message (ACRequest) for transmission to the SS. In response to the Active Call Request message, SS sends an acknowledgement message (ACK) to SC. The SS then sends an Active Call Request message (ACRequest) to RS, and RS responds by sending an acknowledgement message (ACK) to SS. The RS also sends an Active Call Request message (ACRequest) to RC. If RC is able to process the request message (ACRequest) from RS, then RC sends an acknowledgement message (ACK) to RS. During this Active Call Request call flow, if any Active Call Request message (ACRequest) remains undelivered and expiry is reached, then SC, SS (optional) and RS (optional) mark the status of the request as "failed to deliver" and each processing endpoint in the chain ceases to forward the request any further.

Continuing with the active call flow following the Active Call Request, the SC Active Call Response is a message generated on the handset by the recipient of the Active Call Request in response to the Active Call Request described above. The sender of an Active Call Request is the recipient of the Active Call Response, and the recipient of the Active Call Request is the sender of Active Call Response.

In response to receipt of the Active Call Request, and with continuing reference to **Figure 14**, RC generates an Active Call response message (ACResponse). In response to receipt of the Active Call Request message (ACRequest), RS sends an acknowledgement message (ACK) to RC. In response to receipt of the acknowledgement message (ACK) RS sends an Active Call Response message (ACResponse) to SS. During the Active Call Response described herein if RC chooses to ignore the response message (ACResponse) and does not respond with a response message , then SC, SS, RS, and RC expire the response and set the status as "expired".

In response to receipt of the Active Call Response message (ACResponse) SS sends an acknowledgement message (ACK) to RS. In response to receipt of the acknowledgement message SS sends an Active Call Response message (ACResponse) to SC. During the Active Call Response described herein if RC chooses to ignore the Active Call Response message (ACResponse) and does not respond, then SC, SS, RS, and RC expire the response and set the status as "expired".

In response to receipt of the Active Call Response message (ACResponse) SC sends an acknowledgement message (ACK) to SS. During the Active Call Response if RC chooses to ignore the Active Call Response message (ACResponse) and does not respond, then SC, SS, RS, and RC expire the response and set the status as "expired". Furthermore, if the status of the Active Call Response message (ACResponse) remains "undelivered" and expiry of the Active Call Request message (ACRequest) is reached, then RC, RS, SS and SC mark the Active Call Request message (ACRequest) status as "expired"; if the Active Call Response message (ACResponse) status is "undelivered" and the status of the Active Call Request message (ACRequest) changes to "expired", an endpoint marks the Active Call Response message (ACResponse) as "expired".

Continuing with the active call flow following the Active Call Response, delivery confirmations are sent after both Active Call Requests and Active Call Responses. The sender of an active call request is the recipient of the active call response, and the recipient of the active call request is the sender of active call response.

In response to an Active Call Request, and with continuing reference to **Figure 14**, RC sends a Delivery Confirmation message to SC, where RC is connected to RS (Recipient Server) and SC is connected to SS (Sender Server). A similar Delivery Confirmation message in response to the Active Call Response message is also possible. In response to receipt of the Delivery Confirmation message RS sends an

acknowledgement message (ACK) to RC. In response to receipt of the acknowledgement message (ACK) RS sends a Delivery Confirmation message to SS. In response to receipt of the Delivery Confirmation message SS sends an acknowledgement message (ACK) to RS. In response to receipt of the acknowledgement message (ACK) SS sends a Delivery Confirmation message to SC. In response to receipt of the delivery confirmation message SC sends an acknowledgement message (ACK) to SS

Some examples follow of alternative AMC system configurations that include the facilitator and client described above. **Figure 15** is a block diagram of an AMC system 1500, under an alternative embodiment. The AMC system 1500 includes a server or other processor-based device hosting the facilitator 102. The facilitator 102 communicates with one or more client devices 101 to provide AMC system functions among the client devices 101 via network couplings that include the Internet 104a and a telecommunications network 104b. The telecommunications network 104b includes, for example, a cellular telephone network or a public switched telephone network (PTSN), but can be other voice and data communication networks as known in the art. The cellular telephone network can use communication protocols that include, for example, Global System for Mobile communication (GSM), General Packet Radio Service (GPRS), Code Division Multiple Access (CDMA), and Time Division Multiple Access (TDMA), but are not so limited.

**Figure 15** is a block diagram of an AMC system 1600, under another alternative embodiment. The AMC system 1600 includes a server hosting the facilitator 102, and the facilitator 102 communicates with one or more client devices 101 to provide AMC system functions among the client devices 101 via network couplings that include the Internet 104a and/or multiple telecommunications networks 104b1 to 104bn. The telecommunications networks 104b1-104bn are as described above with reference to **Figure 15**, but are not so limited.

**Figure 17** is a block diagram of an AMC system 1700, under yet another alternative embodiment. The AMC system 1700 includes a server hosting the facilitator 102, and the server/facilitator 102 is a component of a telecommunications network operator infrastructure. The facilitator 102 communicates with one or more client devices 101 to provide AMC system functions among the client devices 101 via network couplings 104, as described above, but is not so limited.

**Figure 18** is a block diagram of an AMC system 1800 in an enterprise domain, under another alternative embodiment. The AMC system 1800 includes a server hosting the facilitator 102 where the server/facilitator 102 is a component of a corporate or enterprise infrastructure 1802. The server can host numerous additional applications 1806 in addition to the facilitator 102 or can be dedicated to the facilitator 102. The facilitator 102 communicates with one or more client devices 101 in the public domain 1804 to provide AMC system functions among the client devices 101 via network couplings 104. The network couplings 104 include, for example, the Internet and one or more telecommunication service provider infrastructures, but can include any number/type of couplings. The facilitator 102 also communicates with one or more client devices 101E in the enterprise domain 1802 to provide AMC system functions among the client devices 101E as described below. The client devices 101E in the enterprise domain 1802 are shown coupled to one or more LANs, but are not so limited.

**Figure 19** is a block diagram of an AMC system 1950 in a public domain coupled across components of an enterprise domain, under another alternative embodiment. The AMC system 1950 includes a server hosting the facilitator 102 where the server/facilitator 102 is a component of a carrier or service provider infrastructure or hosted data center infrastructure for example, but is not so limited. The facilitator 102 communicates with one or more client devices 101 in the public domain 1904 to provide AMC system functions among the client devices 101 via network couplings 104. The network couplings 104 include, for example, the Internet and one or more telecommunication service provider infrastructures, but can include any number/type of couplings. The facilitator 102 also communicates with components of the enterprise domain 1902 including, for example, one or more client devices 101E, one or more enterprise servers 1908, and one or more LANs. The facilitator 102 provides AMC system functions among the client devices 101E as described below. The client devices 101E in the enterprise domain 1902 are shown coupled to one or more LANs, but are not so limited.

As an alternative to the couplings of this AMC system 1900, the facilitator can be hosted on one or more servers (not shown) of the telecommunications network operator. The facilitator of the telecommunications network operator couples to the enterprise servers via local contact servers (not shown) and/or Virtual Private Network (VPN) couplings, but is not so limited.

**Figure 20** is a block diagram of an AMC system 2000 in an enterprise domain, under still another alternative embodiment. The AMC system 2000 includes one or more facilitators that form facilitator clusters 602a and 602b within each of a number of enterprise domains 603a and 603b. Facilitators of the facilitator clusters 602a and 602b communicate with one or more client devices 101 to provide AMC system functions among the client devices 101 via network couplings 104. The network couplings 104 include, for example, at least one of the Internet and multiple telecommunication service providers 604a and 604b, but can include any number/type of couplings. The facilitators also couple with at least one of corporate directory servers and/or electronic mail (email) servers 610a/610b, authentication servers 612a/612b, and management consoles 614a/614b of the enterprise domains 603a/603b, but are not so limited.

**Figure 21** is a block diagram of an active mobile collaboration (AMC) system 2100, under an embodiment. The AMC system 2100 includes any number X(n) of communication devices 101 coupled for communication via one or more facilitators 102 and one or more couplings 104. One or more of the communication devices 101 include an AMC client application. Additionally, one or more of the communication devices 101 include the facilitator 102. The AMC client applications and facilitator applications function to allow users of the communication devices to dynamically manage how and when mobile calls take place, intelligently screen calls based on caller identity, urgency, and subject matter, determine which contacts in a directory are available to talk and which ones choose not to be disturbed, and increase accessibility of enterprise and personal contact information from mobile phones, as described in detail below.

The AMC system components including the facilitator and AMC client described above function to allow users of the client devices or handsets like cellular telephones to quickly coordinate conversations, screen unwanted calls and interruptions and access enterprise directories. Specifically, the AMC system components increase call success rates by dynamically managing how and when mobile calls take place, let users intelligently screen calls based on caller identity, urgency and subject matter, quickly show which contacts are available to talk and which contacts choose not to be disturbed, reduce interruptions while encouraging urgently needed call-backs, and increase accessibility of enterprise and personal contact information from mobile phones.

The communication systems of an embodiment include a method comprising receiving information of a voicemail message from a voicemail system that is a server-based system of an enterprise. The voicemail message is received at the voicemail system. The method of an embodiment includes automatically synchronizing a state of the voicemail message between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

Automatically synchronizing a state of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to one or more of access of the voicemail message from the mobile communication device, access of the voicemail message from the access component, playback of the voicemail message from the mobile communication device, playback of the voicemail message from the access component, deletion of the voicemail message from the mobile communication device, and deletion of the voicemail message from the access component.

The method of an embodiment includes transmitting notification of the voicemail message to the mobile communication device.

The method of an embodiment includes transmitting notification of the voicemail message to the access component.

The notification of an embodiment includes initiating a conversation thread at the mobile communication device.

The notification of an embodiment includes an option to playback the voicemail message at the mobile communication device.

The playback of an embodiment includes downloading to the mobile communication device an audio file of the voicemail message. The playback of an embodiment includes playback of the voicemail message at the mobile communication device using the downloaded audio file.

The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to the playback at the mobile communication device.

The playback of an embodiment includes a synchronous request-response file download mechanism between the mobile communication device and the voicemail system via a facilitator.

The facilitator of an embodiment receives a request for the voicemail message from the mobile communication device. The facilitator of an embodiment fetches the voicemail message from the voicemail system. The facilitator of an embodiment transfers an audio file including information of the voicemail message to the mobile communication device.

The method of an embodiment includes converting the audio file from a first format to a second format.

A client of the mobile communication device of an embodiment is configured to receive the audio file. The client of the mobile communication device of an embodiment is configured to store the audio file. The client of the mobile communication device of an embodiment is configured to generate an interface by which the audio file can be accessed for an operation. The operation includes one or more of playing, pausing, and rewinding the voicemail message.

The downloading of an embodiment includes downloading one or more of a partial voicemail message, a single voicemail message, and a plurality of voicemail messages.

The method of an embodiment includes storing on the mobile communication device one or more of the partial voicemail message, the single voicemail message, and the plurality of voicemail messages.

The downloading of an embodiment includes one or more of automatic downloading, downloading in response to user action, and downloading in response to a parameter. The parameter of an embodiment includes one or more of importance of the voicemail message, an originator of the voicemail message, and a network parameter of a communication network providing communications to the mobile communication device.

The playback of an embodiment includes fetching the voicemail message from the voicemail system. The playback of an embodiment includes storing the voicemail message at a facilitator.

The method of an embodiment includes generating at a client of the mobile communication device a command for the facilitator to link a telephone number to the voicemail message. The method of an embodiment includes transferring the telephone number to the client in response to the command.

The method of an embodiment includes initiating a telephone call between the client and the facilitator using the telephone number. The method of an embodiment

includes transmitting the voicemail message from the facilitator to the client via the telephone call.

The method of an embodiment includes generating at a client of the mobile communication device a request to play the voicemail message. The method of an embodiment includes the facilitator initiating a telephone connection with the client in response to the request to play. The method of an embodiment includes the facilitator playing the voicemail message over the telephone connection.

The method of an embodiment includes receiving a delete input at the mobile communication device. The method of an embodiment includes deleting the voicemail message at the mobile communication device and the voicemail system in response to the delete input. The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to the delete input.

The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to access of the voicemail message from the mobile communication device.

The notification of an embodiment includes providing one or more of audio and visual indicators at the access component.

The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to playback of the voicemail message initiated at the telephone.

The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to access of the voicemail message initiated at the telephone.

The notification of an embodiment includes an option to automatically respond to the voicemail message from the mobile communication device.

The automatic response of an embodiment includes generation and transmission of an electronic message to the originator of the voicemail message.

The electronic message of an embodiment includes one or more of an electronic mail message, a text message, an instant message, a Short Message Service (SMS) message, a Multimedia Messaging Service (MMS) message, and an Enhanced Messaging Service (EMS) message.

The automatic response of an embodiment includes generation and transmission of a telephone call to the originator of the voicemail message.

The automatic response of an embodiment includes generating a call request to the originator of the voicemail message.

Generating the call request of an embodiment includes transferring the call request from the mobile communication device to a target device of the originator via a data coupling.

The method of an embodiment includes providing a delivery confirmation to the mobile communication device in response to delivery of the transferred call request to the target device.

The method of an embodiment includes generating a plurality of response options on the target device in response to receipt of the call request. The response options include at least one of accepting the call request, postponing the call request, generating a return data message including at least one of a text message and a voice message to the mobile communication device, and initiating the voice call.

The method of an embodiment includes generating a response message in response to selection of a response option at the target device and transferring the response message to the mobile communication device over the data coupling.

The method of an embodiment includes receiving a delete input at the access component. The method of an embodiment includes deleting the voicemail message at the mobile communication device and the voicemail system in response to the delete input. The method of an embodiment includes changing a state of the voicemail message at the mobile communication device and the access component in response to the delete input.

The method of an embodiment includes determining an identification of an originator of the voicemail message.

The method of an embodiment includes mapping originator information of the originator to an integrated directory, wherein the integrated directory includes contacts representing a set of contacts of a plurality of directories of the mobile communication device and the enterprise.

The mobile communication device of an embodiment operates external to the enterprise.

The mobile communication device of an embodiment operates internal to the enterprise.

The mobile communication device of an embodiment is one or more of an Internet Protocol (IP) telephone, WiFi device, cellular telephone, multi-mode

telephone, wireless device, personal digital assistant, and Worldwide Interoperability for Microwave Access (WiMAX) device.

The access component of an embodiment includes a telephone coupled to a private branch exchange (PBX).

The access component of an embodiment includes a telephone coupled to a central exchange (Centrex).

The method of an embodiment includes displaying the information of the voicemail message on the mobile communication device.

The method of an embodiment includes displaying the state of the voicemail message on the mobile communication device.

The method of an embodiment includes displaying a voicemail control interface on the mobile communication device by which the audio file can be accessed for an operation. The operation of an embodiment includes one or more of playing, pausing, rewinding, fast-forwarding, deleting, and changing the state of the voicemail message. The operation of an embodiment includes responding to an originator of the voicemail message, wherein the responding includes one or more of generating and transmitting a telephone call, call request, electronic mail message, text message, instant message (IM), voice IM, Short Message Service (SMS) message, Multimedia Messaging Service (MMS) message, and Enhanced Messaging Service (EMS) message.

The method of an embodiment includes displaying information of the voicemail message of the enterprise on the client device. The information of the voicemail message of an embodiment includes one or more of an originator of the voicemail message, a length of the voicemail message, and a received time of the voicemail message.

The displaying of an embodiment is automatically performed in response to receipt of the voicemail message at the enterprise.

The method of an embodiment includes changing the state of the voicemail message at the mobile communication device to an expired state after a first period of time has elapsed following transmission to the mobile communication device of a notification of the voicemail message.

The method of an embodiment includes displaying the voicemail message as an expired conversation at the mobile communication device.

The notification of an embodiment includes an option to playback the voicemail message at the mobile communication device. The playback of the voicemail message

of an embodiment in an expired state can be initiated using information of the notification.

The method of an embodiment includes deleting the notification from the mobile communication device after a second period of time has elapsed following transmission of the notification.

The communication systems of an embodiment include a method comprising transmitting voicemail information of a voicemail message to a client device. The voicemail message is received at a voicemail system that is a server-based system of an enterprise. The method of an embodiment includes displaying the voicemail information at the client device. The method of an embodiment includes automatically synchronizing a state of the voicemail information between the client device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

The communication systems of an embodiment include a method comprising receiving a voicemail message in a voicemail system. The voicemail system is a server-based system of an enterprise. The method of an embodiment includes transmitting notification of the voicemail message to a remote device and an access device of the voicemail system. The method of an embodiment includes automatically maintaining an accurate state of the message at the remote device and the access device. The accurate state reflects changes in the voicemail message made via one or more of the remote device and components of the voicemail server system.

The communication systems of an embodiment include a system comprising a voicemail system coupled to an enterprise and configured to receive and store a voicemail message. The system of an embodiment includes a server coupled to at least one communication network. The system of an embodiment includes at least one adapter coupled to the server and the voicemail system. The adapter is configured to automatically synchronize a state of the voicemail message between a mobile communication device external to the enterprise and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

The adapter of an embodiment is coupled to at least one database of the voicemail system, wherein the adapter is configured for a type of the database.

The database of an embodiment is coupled to a voicemail server of the enterprise.

The database of an embodiment is coupled to a plurality of voicemail servers of the enterprise.

The voicemail servers of an embodiment comprise a plurality of voicemail configurations.

The adapter of an embodiment is coupled to at least one component of the server. The at least one component of an embodiment is one or more of a calendar component, an electronic mail component, and a contacts component. The at least one component of an embodiment is a voicemail component that is configured for one or more of a type and configuration of the voicemail system. The voicemail component of an embodiment is independent of the voicemail system.

The at least one adapter of an embodiment includes a plurality of adapters.

The adapters of an embodiment include a first set of adapters coupled to a voicemail component of the server. The voicemail component of an embodiment is configured for one or more of a type and configuration of the voicemail system. The voicemail component of an embodiment is independent of the voicemail system.

Each adapter of the first set of an embodiment is coupled a database of the voicemail system. The adapter of an embodiment is configured for a type of the database to which it is coupled.

The database of an embodiment is coupled to a voicemail server of the enterprise.

The database of an embodiment is coupled to a plurality of voicemail servers of the enterprise. The plurality of voicemail servers of an embodiment comprises a plurality of voicemail configurations.

An adapter of the first set of an embodiment is coupled to a proprietary database of the voicemail system, wherein the proprietary database is proprietary to a voicemail server to which it is coupled, wherein the adapter is configured to interface with the proprietary database. The adapter of an embodiment is configured to support Internet Message Access Protocol (IMAP) version 4 (IMAP4).

An adapter of the first set of an embodiment is connected to a voicemail server of the voicemail system, wherein the adapter is a proprietary adapter that is proprietary to a voicemail server to which it is coupled. The voicemail server of an embodiment includes an embedded database.

The system of an embodiment includes a second adapter coupled to the server and one or more of a calendar and contacts system of the enterprise. The second adapter of an embodiment is configured to automatically synchronize a state of calendar and contacts information between a mobile communication device external to the enterprise and an access component of the enterprise in response to changes in the calendar and contacts information made via one or more of the mobile communication device and the access component. The second adapter of an embodiment is coupled to a calendar and contacts component of the server.

The calendar and contacts component of the server of an embodiment is configured for one or more of a type and configuration of the calendar and contacts system of the enterprise. The calendar and contacts component of an embodiment is independent of the calendar and contacts system of the enterprise.

The communication systems of an embodiment include a computer readable media including executable instructions which, when executed in a processing system, provides visual voicemail by receiving information of a voicemail message from a voicemail system that is a server-based system of an enterprise. The voicemail message is received at the voicemail system. The instructions of an embodiment, when executed, automatically synchronize a state of the voicemail message between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

Aspects of the communications systems described herein may be implemented as functionality programmed into any of a variety of circuitry, including programmable logic devices (PLDs), such as field programmable gate arrays (FPGAs), programmable array logic (PAL) devices, electrically programmable logic and memory devices and standard cell-based devices, as well as application specific integrated circuits (ASICs). Some other possibilities for implementing aspects of the communications systems include: microcontrollers with memory (such as electronically erasable programmable read-only memory (EEPROM)), embedded microprocessors, firmware, software, etc. Furthermore, aspects of the communications systems may be embodied in microprocessors having software-based circuit emulation, discrete logic (sequential and combinatorial), custom devices, fuzzy (neural) logic, quantum devices, and hybrids of any of the above device types. Of course the underlying device technologies may be provided in a variety of component types, e.g., metal-oxide semiconductor field-effect

transistor (MOSFET) technologies like complementary metal-oxide semiconductor (CMOS), bipolar technologies like emitter-coupled logic (ECL), polymer technologies (e.g., silicon-conjugated polymer and metal-conjugated polymer-metal structures), mixed analog and digital, etc.

It should be noted that components of the various systems and methods disclosed herein may be described using computer aided design tools and expressed (or represented), as data and/or instructions embodied in various computer-readable media, in terms of their behavioral, register transfer, logic component, transistor, layout geometries, and/or other characteristics. Computer-readable media in which such formatted data and/or instructions may be embodied include, but are not limited to, non-volatile storage media in various forms (e.g., optical, magnetic or semiconductor storage media) and carrier waves that may be used to transfer such formatted data and/or instructions through wireless, optical, or wired signaling media or any combination thereof.

Examples of transfers of such formatted data and/or instructions by carrier waves include, but are not limited to, transfers (uploads, downloads, e-mail, etc.) over the Internet and/or other computer networks via one or more data transfer protocols (e.g., HTTP, FTP, SMTP, etc.). When received within a computer system via one or more computer-readable media, such data and/or instruction-based expressions of the above described systems and methods may be processed by a processing entity (e.g., one or more processors) within the computer system in conjunction with execution of one or more other computer programs.

Unless the context clearly requires otherwise, throughout the description, the words "comprise," "comprising," and the like are to be construed in an inclusive sense as opposed to an exclusive or exhaustive sense; that is to say, in a sense of "including, but not limited to." Words using the singular or plural number also include the plural or singular number respectively. Additionally, the words "herein," "hereunder," "above," "below," and words of similar import refer to this application as a whole and not to any particular portions of this application. When the word "or" is used in reference to a list of two or more items, that word covers all of the following interpretations of the word: any of the items in the list, all of the items in the list and any combination of the items in the list.

The above description of embodiments of the communications systems is not intended to be exhaustive or to limit the systems and methods described to the precise

form disclosed. While specific embodiments of, and examples for, the communications systems are described herein for illustrative purposes, various equivalent modifications are possible within the scope of other communications systems and methods, as those skilled in the relevant art will recognize. The teachings of the communications systems provided herein can be applied to other processing systems and methods, not only for the systems and methods described above.

The elements and acts of the various embodiments described above can be combined to provide further embodiments. These and other changes can be made to the communications systems in light of the above detailed description.

In general, in the following claims, the terms used should not be construed to limit the communications systems to the specific embodiments disclosed in the specification and the claims, but should be construed to include all systems that operate under the claims. Accordingly, the communications systems are not limited by the disclosure, but instead the scope of the communications systems is to be determined entirely by the claims.

While certain aspects of the communications systems are presented below in certain claim forms, the inventors contemplate the various aspects of the communications systems in any number of claim forms. Accordingly, the inventors reserve the right to add additional claims after filing the application to pursue such additional claim forms for other aspects of the communications systems.

CLAIMS

What is claimed is:

1. A method comprising:

receiving information of a voicemail message from a voicemail system that is a server-based system of an enterprise, wherein the voicemail message is received at the voicemail system; and

automatically synchronizing a state of the voicemail message between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

2. The method of claim 1, wherein automatically synchronizing a state includes changing a state of the voicemail message at the mobile communication device and the access component in response to one or more of access of the voicemail message from the mobile communication device, access of the voicemail message from the access component, playback of the voicemail message from the mobile communication device, playback of the voicemail message from the access component, deletion of the voicemail message from the mobile communication device, and deletion of the voicemail message from the access component.

3. The method of claim 1, comprising transmitting notification of the voicemail message to the mobile communication device.

4. The method of claim 3, comprising transmitting notification of the voicemail message to the access component.

5. The method of claim 4, wherein the notification includes initiating a conversation thread at the mobile communication device.

6. The method of claim 5, wherein the notification includes an option to playback the voicemail message at the mobile communication device.

7. The method of claim 6, wherein the playback includes:

downloading to the mobile communication device an audio file of the voicemail message; and

playback of the voicemail message at the mobile communication device using the downloaded audio file.

8. The method of claim 7, comprising changing a state of the voicemail message at the mobile communication device and the access component in response to the playback at the mobile communication device.

9. The method of claim 7, wherein the playback includes a synchronous request-response file download mechanism between the mobile communication device and the voicemail system via a facilitator.

10. The method of claim 7, wherein the facilitator:

receives a request for the voicemail message from the mobile communication device;

fetches the voicemail message from the voicemail system; and

transfers an audio file including information of the voicemail message to the mobile communication device.

11. The method of claim 7, comprising converting the audio file from a first format to a second format.

12. The method of claim 7, wherein a client of the mobile communication device:

receives the audio file;

stores the audio file; and

generates an interface by which the audio file can be accessed for an operation, wherein the operation includes one or more of playing, pausing, and rewinding the voicemail message.

13. The method of claim 7, wherein the downloading includes downloading one or more of a partial voicemail message, a single voicemail message, and a plurality of voicemail messages.

14. The method of claim 13, comprising storing on the mobile communication device one or more of the partial voicemail message, the single voicemail message, and the plurality of voicemail messages.

15. The method of claim 7, wherein the downloading includes one or more of automatic downloading, downloading in response to user action, and downloading in response to a parameter.

16. The method of claim 15, wherein the parameter includes one or more of importance of the voicemail message, an originator of the voicemail message, and a network parameter of a communication network providing communications to the mobile communication device.

17. The method of claim 6, wherein the playback includes:
fetching the voicemail message from the voicemail system; and
storing the voicemail message at a facilitator.

18. The method of claim 17, comprising:
generating at a client of the mobile communication device a command for the facilitator to link a telephone number to the voicemail message; and
transferring the telephone number to the client in response to the command.

19. The method of claim 17, comprising:
initiating a telephone call between the client and the facilitator using the telephone number;
transmitting the voicemail message from the facilitator to the client via the telephone call.

20. The method of claim 17, comprising:
generating at a client of the mobile communication device a request to play the voicemail message;
the facilitator initiating a telephone connection with the client in response to the request to play;
the facilitator playing the voicemail message over the telephone connection.

21. The method of claim 7, comprising:

receiving a delete input at the mobile communication device;

deleting the voicemail message at the mobile communication device and the voicemail system in response to the delete input; and

changing a state of the voicemail message at the mobile communication device and the access component in response to the delete input.

22. The method of claim 5, comprising changing a state of the voicemail message at the mobile communication device and the access component in response to access of the voicemail message from the mobile communication device.

23. The method of claim 5, wherein the notification includes providing one or more of audio and visual indicators at the access component.

24. The method of claim 23, comprising changing a state of the voicemail message at the mobile communication device and the access component in response to playback of the voicemail message initiated at the telephone.

25. The method of claim 23, comprising changing a state of the voicemail message at the mobile communication device and the access component in response to access of the voicemail message initiated at the telephone.

26. The method of claim 5, wherein the notification includes an option to automatically respond to the voicemail message from the mobile communication device.

27. The method of claim 26, wherein the automatic response includes generation and transmission of an electronic message to the originator of the voicemail message.

28. The method of claim 27, wherein the electronic message includes one or more of an electronic mail message, a text message, an instant message, a Short Message Service (SMS) message, a Multimedia Messaging Service (MMS) message, and an Enhanced Messaging Service (EMS) message.

29. The method of claim 26, wherein the automatic response includes generation and transmission of a telephone call to the originator of the voicemail message.

30. The method of claim 26, wherein the automatic response includes generating a call request to the originator of the voicemail message.

31. The method of claim 30, wherein generating the call request includes transferring the call request from the mobile communication device to a target device of the originator via a data coupling.

32. The method of claim 31, comprising providing a delivery confirmation to the mobile communication device in response to delivery of the transferred call request to the target device.

33. The method of claim 32, comprising generating a plurality of response options on the target device in response to receipt of the call request, wherein the response options include at least one of accepting the call request, postponing the call request, generating a return data message including at least one of a text message and a voice message to the mobile communication device, and initiating the voice call.

34. The method of claim 33, comprising generating a response message in response to selection of a response option at the target device and transferring the response message to the mobile communication device over the data coupling.

35. The method of claim 4, comprising:

receiving a delete input at the access component;

deleting the voicemail message at the mobile communication device and the voicemail system in response to the delete input; and

changing a state of the voicemail message at the mobile communication device and the access component in response to the delete input.

36. The method of claim 1, comprising determining an identification of an originator of the voicemail message.

37. The method of claim 36, comprising mapping originator information of the originator to an integrated directory, wherein the integrated directory includes contacts representing a set of contacts of a plurality of directories of the mobile communication device and the enterprise.

38. The method of claim 1, wherein the mobile communication device operates external to the enterprise.

39. The method of claim 1, wherein the mobile communication device operates internal to the enterprise.

40. The method of claim 1, wherein the mobile communication device is one or more of an Internet Protocol (IP) telephone, WiFi device, cellular telephone, multi-mode telephone, wireless device, personal digital assistant, and Worldwide Interoperability for Microwave Access (WiMAX) device.

41. The method of claim 1, wherein the access component includes a telephone coupled to a private branch exchange (PBX).

42. The method of claim 1, wherein the access component includes a telephone coupled to a central exchange (Centrex).

43. The method of claim 1, comprising displaying the information of the voicemail message on the mobile communication device.

44. The method of claim 1, comprising displaying the state of the voicemail message on the mobile communication device.

45. The method of claim 1, comprising displaying a voicemail control interface on the mobile communication device by which the audio file can be accessed for an operation.

46. The method of claim 45, wherein the operation includes one or more of playing, pausing, rewinding, fast-forwarding, deleting, and changing the state of the voicemail message.

47. The method of claim 45, wherein the operation includes responding to an originator of the voicemail message, wherein the responding includes one or more of generating and transmitting a telephone call, call request, electronic mail message, text message, instant message (IM), voice IM, Short Message Service (SMS) message, Multimedia Messaging Service (MMS) message, and Enhanced Messaging Service (EMS) message.

48. The method of claim 1, comprising displaying information of the voicemail message of the enterprise on the client device, the information of the voicemail message including one or more of an originator of the voicemail message, a length of the voicemail message, and a received time of the voicemail message.

49. The method of claim 48, wherein the displaying is automatically performed in response to receipt of the voicemail message at the enterprise.

50. The method of claim 1, comprising changing the state of the voicemail message at the mobile communication device to an expired state after a first period of time has elapsed following transmission to the mobile communication device of a notification of the voicemail message.

51. The method of claim 50, comprising displaying the voicemail message as an expired conversation at the mobile communication device.

52. The method of claim 50, wherein the notification includes an option to playback the voicemail message at the mobile communication device, wherein the playback of the voicemail message in an expired state can be initiated using information of the notification.

53. The method of claim 50, comprising deleting the notification from the mobile communication device after a second period of time has elapsed following transmission of the notification.

54. A method comprising:

transmitting voicemail information of a voicemail message to a client device, wherein the voicemail message is received at a voicemail system that is a server-based system of an enterprise;

displaying the voicemail information at the client device; and

automatically synchronizing a state of the voicemail information between the client device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

55. A method comprising:

receiving a voicemail message in a voicemail system, wherein the voicemail system is a server-based system of an enterprise;

transmitting notification of the voicemail message to a remote device and an access device of the voicemail system; and

automatically maintaining an accurate state of the message at the remote device and the access device, wherein the accurate state reflects changes in the voicemail message made via one or more of the remote device and components of the voicemail server system.

56. A system comprising:

a voicemail system coupled to an enterprise and configured to receive and store a voicemail message;

a server coupled to at least one communication network; and

at least one adapter coupled to the server and the voicemail system, wherein the at least one adapter is configured to automatically synchronize a state of the voicemail message between a mobile communication device external to the enterprise and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

57. The system of claim 56, wherein the adapter is coupled to at least one database of the voicemail system, wherein the adapter is configured for a type of the database.

58. The system of claim 57, wherein the database is coupled to a voicemail server of the enterprise.

59. The system of claim 57, wherein the database is coupled to a plurality of voicemail servers of the enterprise.

60. The system of claim 59, wherein the plurality of voicemail servers comprise a plurality of voicemail configurations.

61. The system of claim 57, wherein the adapter is coupled to at least one component of the server.

62. The system of claim 61, wherein the at least one component is one or more of a calendar component, an electronic mail component, and a contacts component.

63. The system of claim 61, wherein the at least one component is a voicemail component that is configured for one or more of a type and configuration of the voicemail system, wherein the voicemail component is independent of the voicemail system.

64. The system of claim 56, wherein the at least one adapter includes a plurality of adapters.

65. The system of claim 64, wherein the plurality of adapters includes a first set of adapters coupled to a voicemail component of the server, wherein the voicemail component is configured for one or more of a type and configuration of the voicemail system, wherein the voicemail component is independent of the voicemail system.

66. The system of claim 65, wherein each adapter of the first set is coupled a database of the voicemail system, wherein the adapter is configured for a type of the database to which it is coupled.

67. The system of claim 66, wherein the database is coupled to a voicemail server of the enterprise.

68. The system of claim 66, wherein the database is coupled to a plurality of voicemail servers of the enterprise.

69. The system of claim 68, wherein the plurality of voicemail servers comprise a plurality of voicemail configurations.

70. The system of claim 65, wherein an adapter of the first set is coupled to a proprietary database of the voicemail system, wherein the proprietary database is proprietary to a voicemail server to which it is coupled, wherein the adapter is configured to interface with the proprietary database.

71. The system of claim 70, wherein the adapter is configured to support Internet Message Access Protocol (IMAP) version 4 (IMAP4).

72. The system of claim 65, wherein an adapter of the first set is connected to a voicemail server of the voicemail system, wherein the adapter is a proprietary adapter that is proprietary to a voicemail server to which it is coupled.

73. The system of claim 72, wherein the voicemail server includes an embedded database.

74. The system of claim 56, comprising a second adapter coupled to the server and one or more of a calendar and contacts system of the enterprise, wherein the second adapter is configured to automatically synchronize a state of calendar and contacts information between a mobile communication device external to the enterprise and an access component of the enterprise in response to changes in the calendar and contacts

information made via one or more of the mobile communication device and the access component.

75. The system of claim 74, wherein the second adapter is coupled to a calendar and contacts component of the server.

76. The system of claim 75, wherein the calendar and contacts component of the server is configured for one or more of a type and configuration of the calendar and contacts system of the enterprise, wherein the calendar and contacts component is independent of the calendar and contacts system of the enterprise.

77. A computer readable media including executable instructions which, when executed in a processing system, provides visual voicemail by:

receiving information of a voicemail message from a voicemail system that is a server-based system of an enterprise, wherein the voicemail message is received at the voicemail system; and

automatically synchronizing a state of the voicemail message between a mobile communication device and an access component of the voicemail system in response to changes in the voicemail message made via one or more of the mobile communication device and the access component.

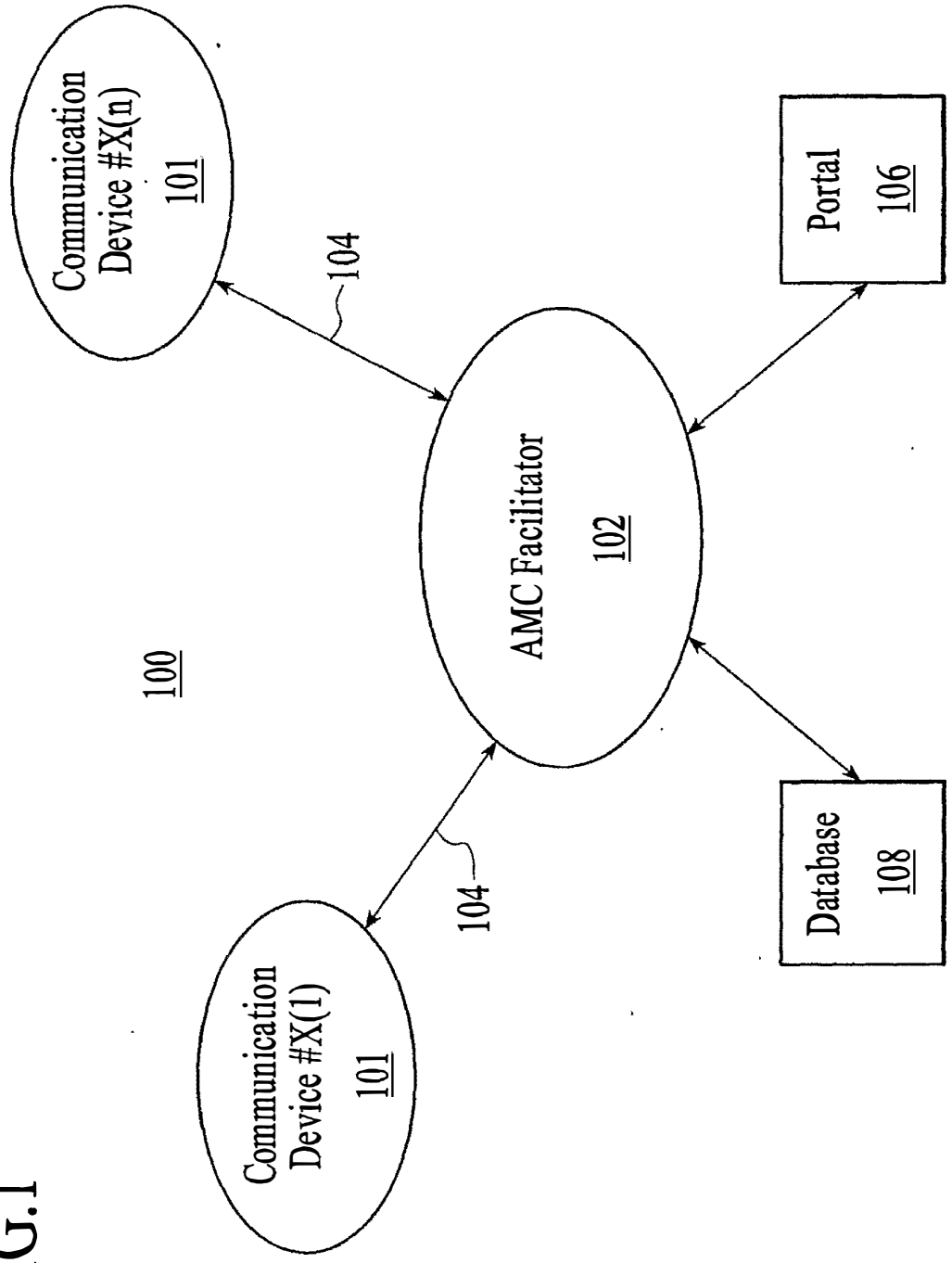
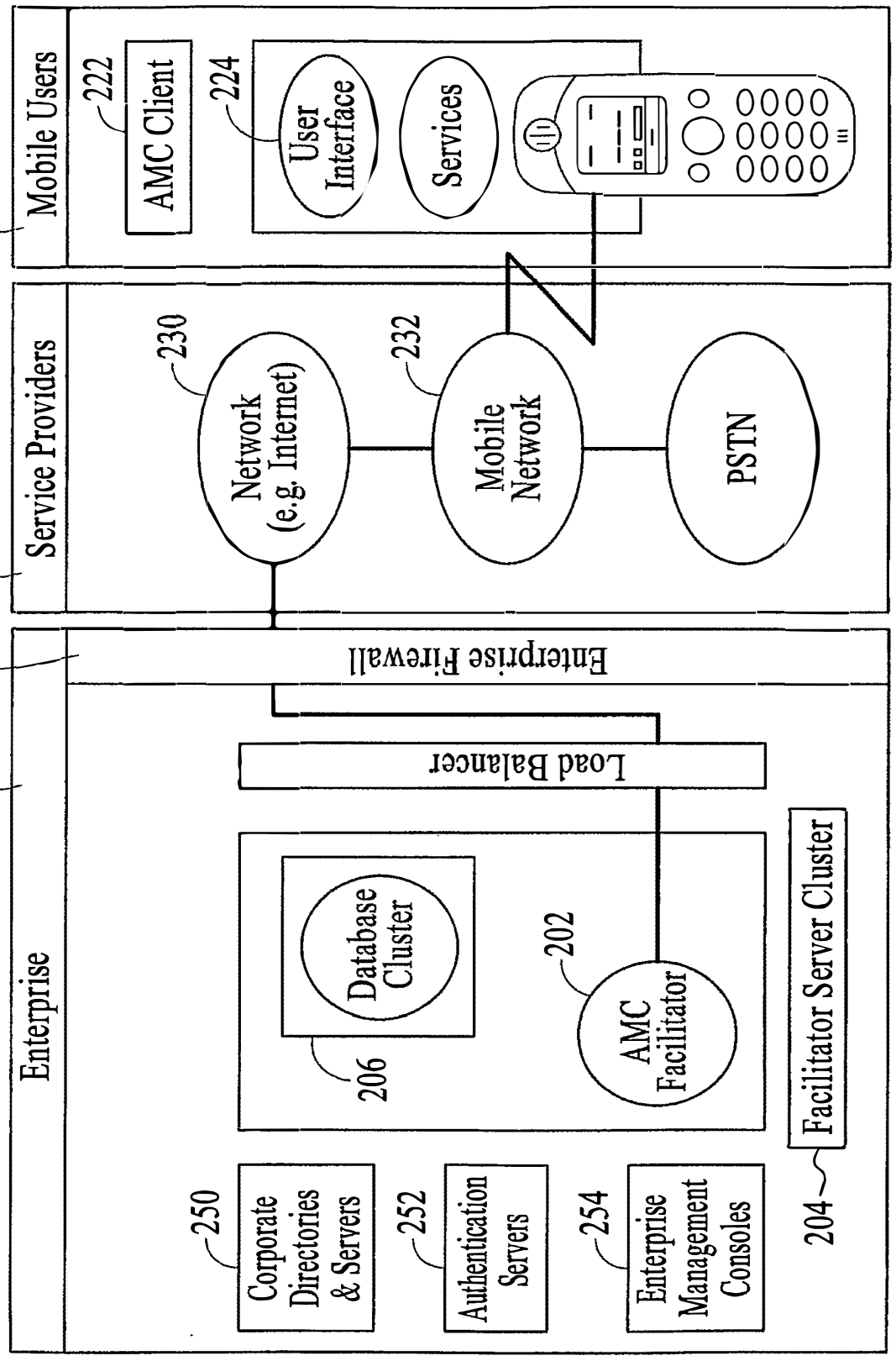

Communication Device #X(n) 101
104
Portal 106
AMC Facilitator 102
100
Database 108
104
Communication Device #X(1) 101

FIG.1

FIG.2


200
200E
200F
200S
200M
Enterprise
Enterprise Firewall
Service Providers
Mobile Users
Load Balancer
Database Cluster
206
AMC Facilitator
202
Facilitator Server Cluster
204
Corporate Directories & Servers
250
Authentication Servers
252
Enterprise Management Consoles
254
Network (e.g. Internet)
230
Mobile Network
232
PSTN
AMC Client
222
User Interface
224
Services

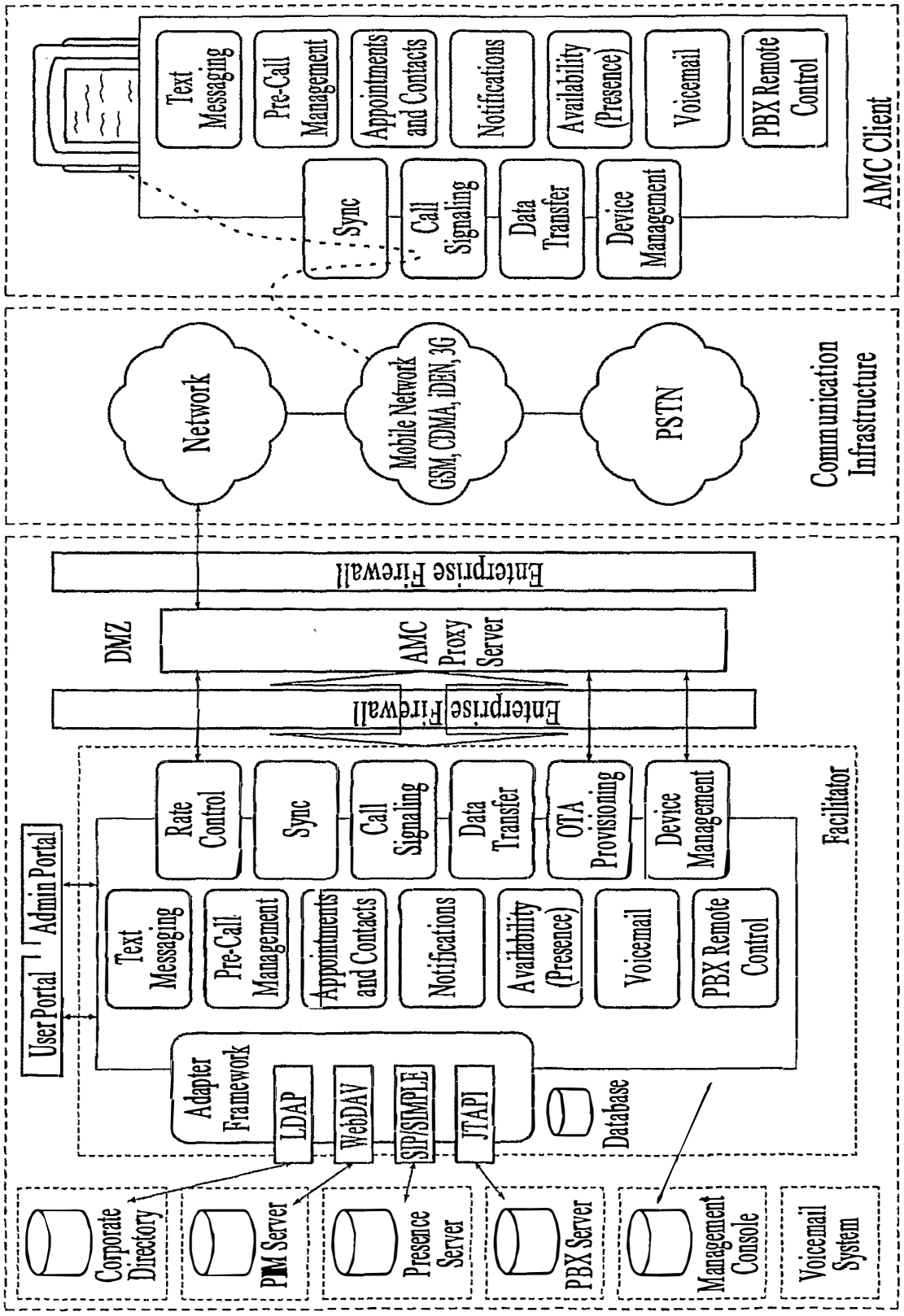
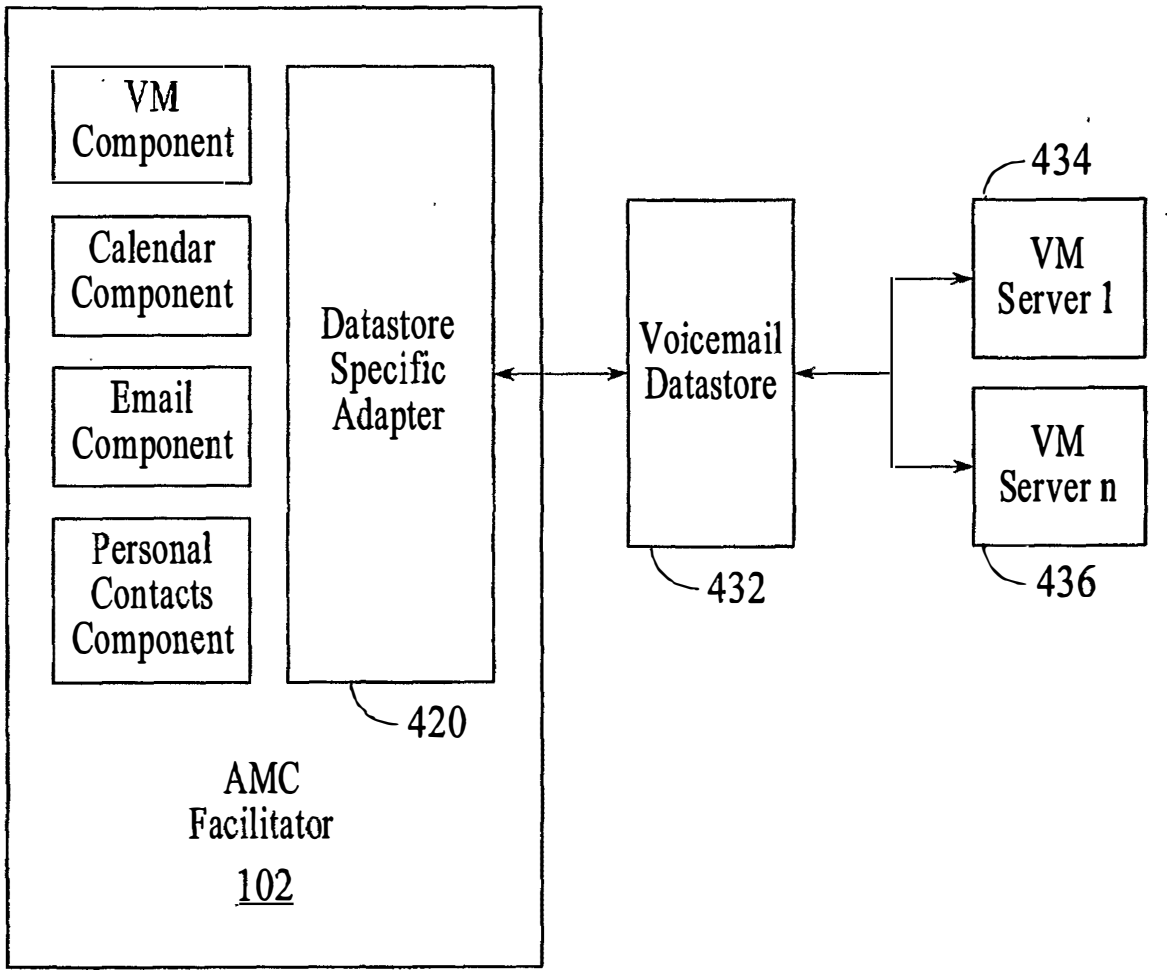
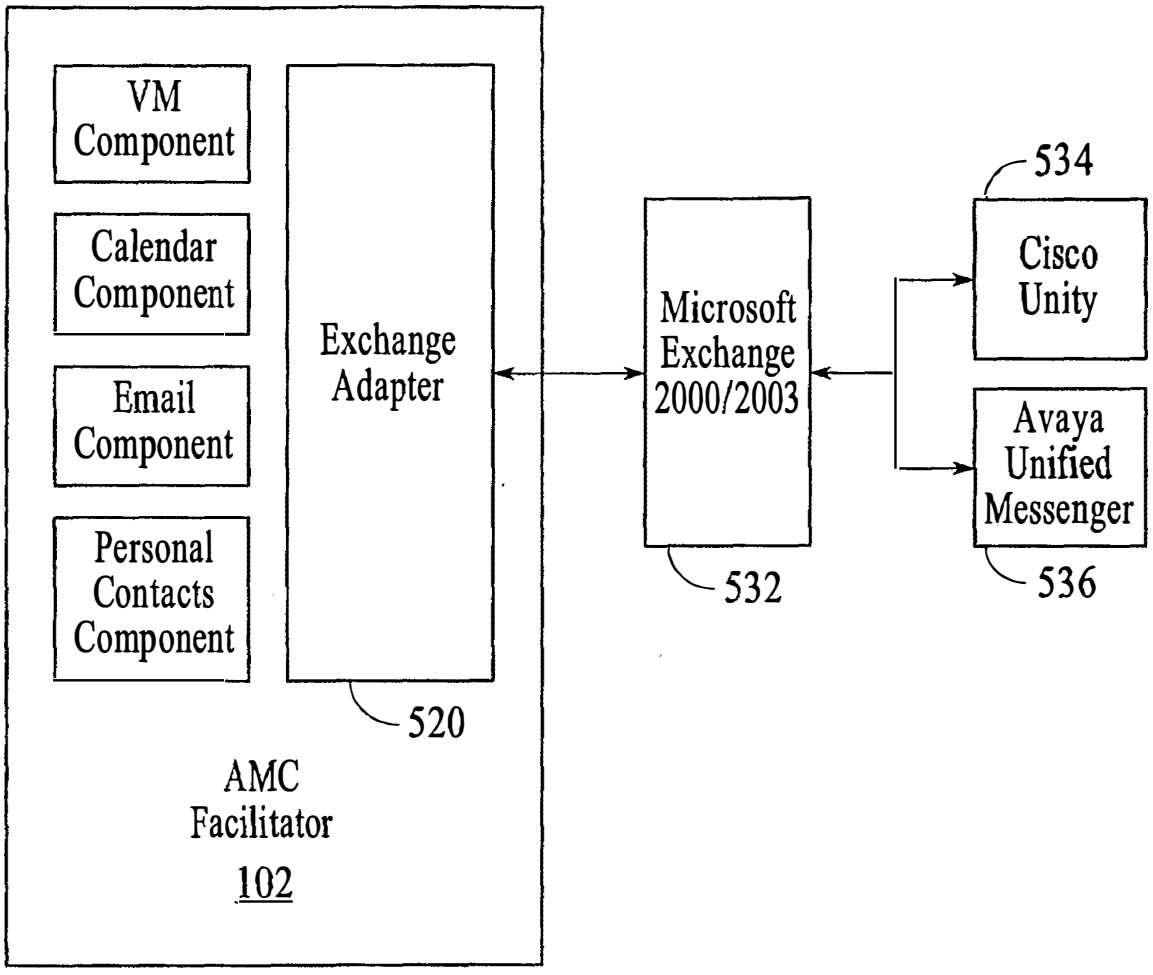

AMC Client
Text Messaging
Pre-Call Management
Appointments and Contacts
Notifications
Availability (Presence)
Voicemail
PBX Remote Control
Sync
Call Signaling
Data Transfer
Device Management
Communication Infrastructure
Network
Mobile Network GSM, CDMA, iDEN, 3G
PSTN
Enterprise Firewall
DMZ
AMC Proxy Server
Enterprise Firewall
Facilitator
User Portal
Admin Portal
Rate Control
Sync
Call Signaling
Data Transfer
OTA Provisioning
Device Management
Text Messaging
Pre-Call Management
Appointments and Contacts
Notifications
Availability (Presence)
Voicemail
PBX Remote Control
Adapter Framework
LDAP
WebDAV
SIP/SIMPLE
JTAPI
Database
Corporate Directory
PIM Server
Presence Server
PBX Server
Management Console
Voicemail System
300

FIG.3


VM Component
Calendar Component
Email Component
Personal Contacts Component
Datastore Specific Adapter
420
AMC Facilitator
102
Voicemail Datastore
432
434
VM Server 1
VM Server n
436

400

FIG.4


VM Component
Calendar Component
Email Component
Personal Contacts Component
Exchange Adapter
520
AMC Facilitator
102
Microsoft Exchange 2000/2003
532
534
Cisco Unity
Avaya Unified Messenger
536

500

FIG.5

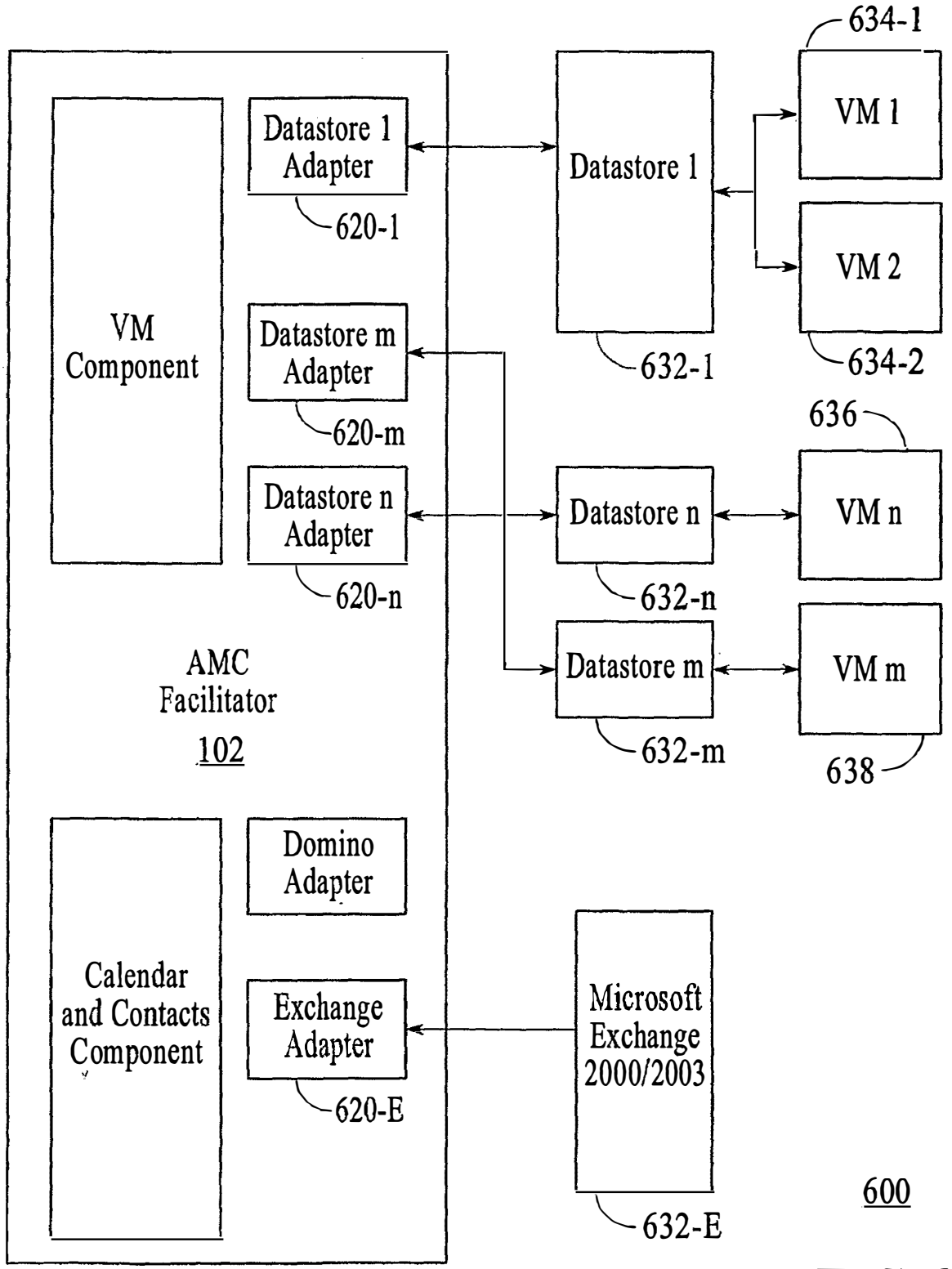

634-1
VM 1
Datastore 1
Datastore 1 Adapter
620-1
VM 2
634-2
VM Component
Datastore m Adapter
620-m
632-1
636
Datastore n Adapter
620-n
Datastore n
VM n
632-n
AMC Facilitator
102
Datastore m
VM m
632-m
638
Domino Adapter
Calendar and Contacts Component
Exchange Adapter
620-E
Microsoft Exchange 2000/2003
632-E
600

FIG.6

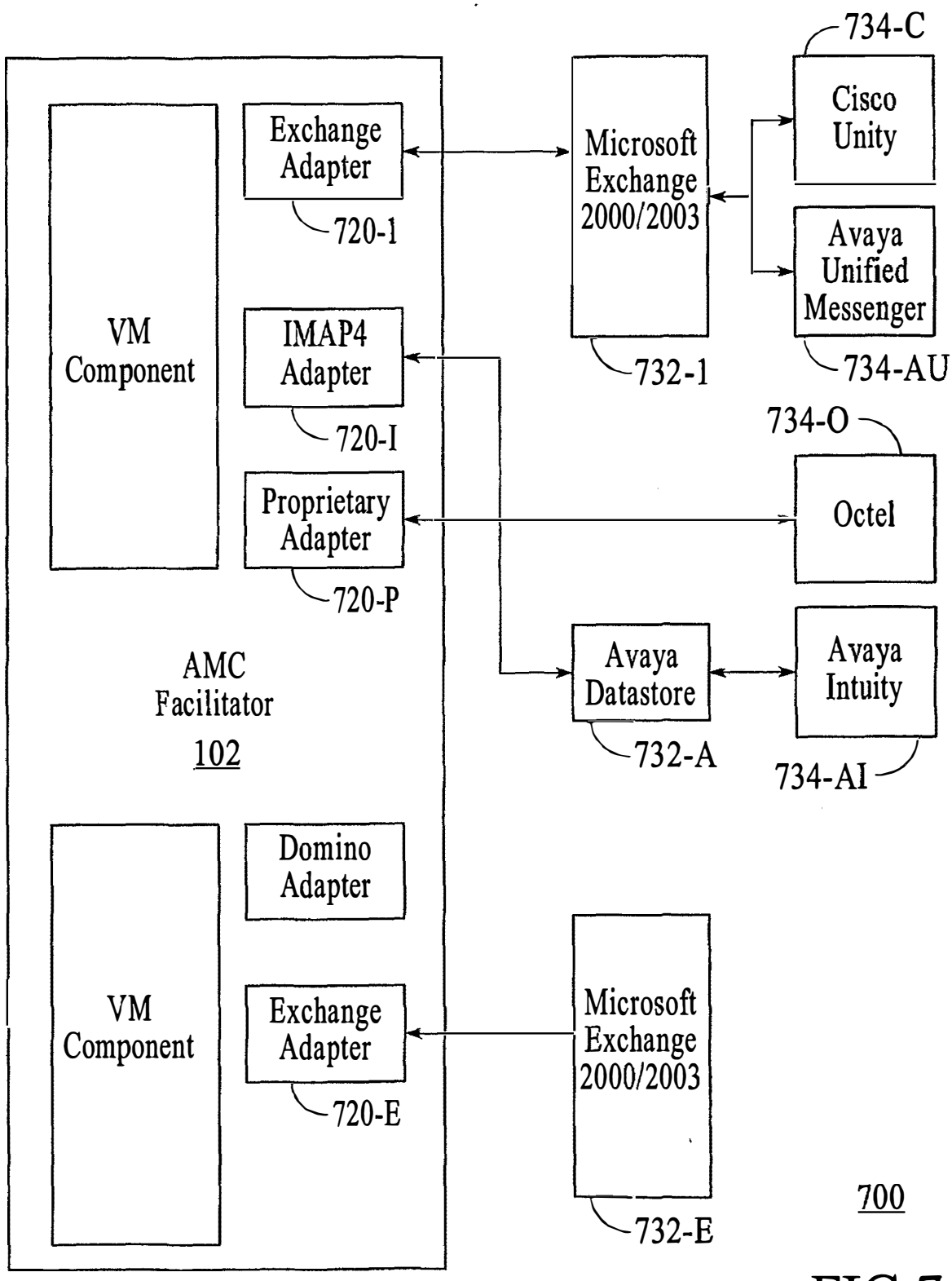

734-C
Cisco Unity
Exchange Adapter
720-1
Microsoft Exchange 2000/2003
Avaya Unified Messenger
VM Component
IMAP4 Adapter
720-I
732-1
734-AU
734-O
Proprietary Adapter
720-P
Octel
AMC Facilitator
102
Avaya Datastore
Avaya Intuity
732-A
734-AI
Domino Adapter
VM Component
Exchange Adapter
720-E
Microsoft Exchange 2000/2003
732-E
700

FIG.7

FIG.8A

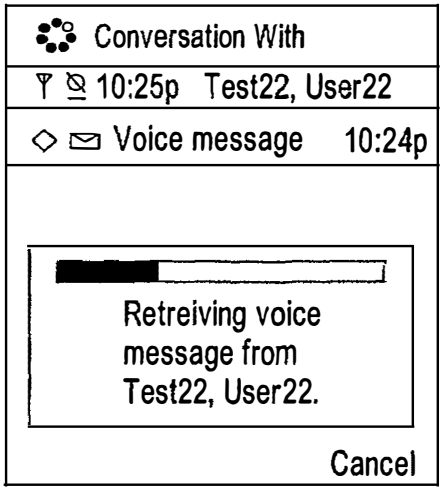
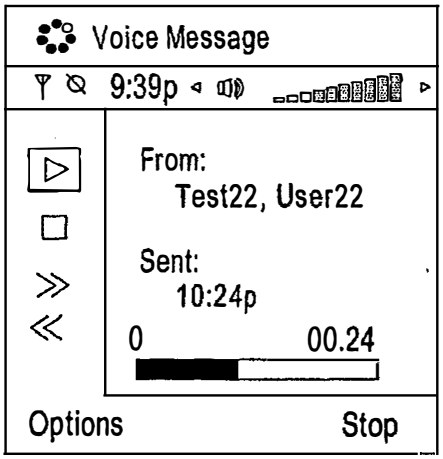

Conversation With
10:25p Test22, User22
Voice message 10:24p
Retreiving voice message from Test22, User22.
Cancel

FIG.8B


Voice Message
9:39p
From:
Test22, User22
Sent:
10:24p
0
00.24
Options
Stop

| Req. | Icon | Title | Description |
|---|---|---|---|
| 1 | ▷ | Play | Starts/resumes playing the voice message from current location.<br>Not visible if already in playing mode. |
| 2 | ⏸ | Pause | Pauses playing at current location.<br>Only visible if in playing mode. |
| 3 | □ | Stop | Stops playing and resets position to start of message.<br>Disabled or dimmed if already stopped. |
| 4 | ≫ | Skip forward | Skips forward 5 seconds or to end of message<br>if less than 5 seconds remaining. |
| 5 | ≪ | Skip backward | Skips backward 5 seconds or to start of message<br>if less than 5 seconds from start of message |
| | | | |

FIG.8C

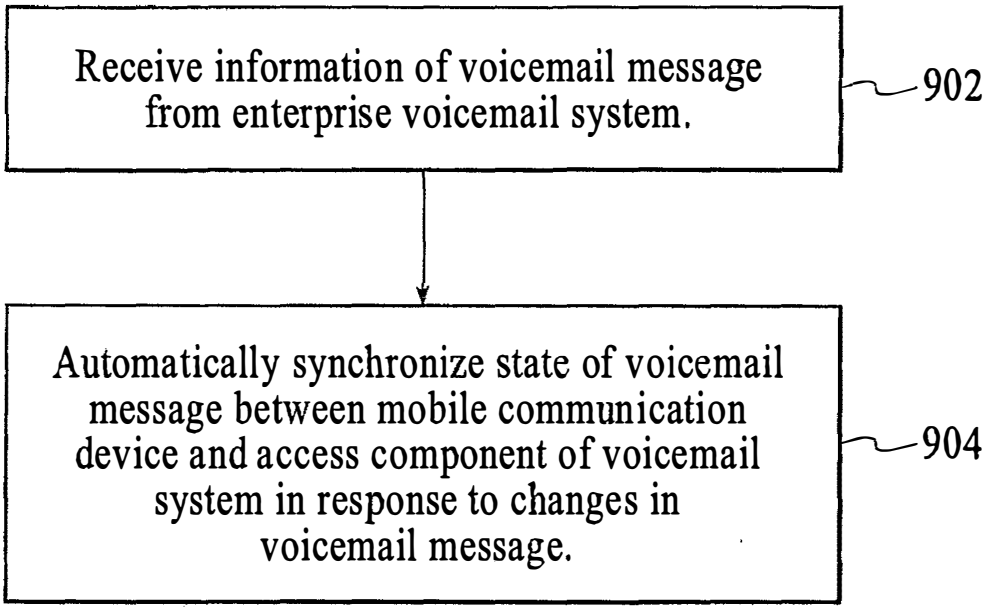

Receive information of voicemail message from enterprise voicemail system.
902
Automatically synchronize state of voicemail message between mobile communication device and access component of voicemail system in response to changes in voicemail message.
904
900

FIG.9

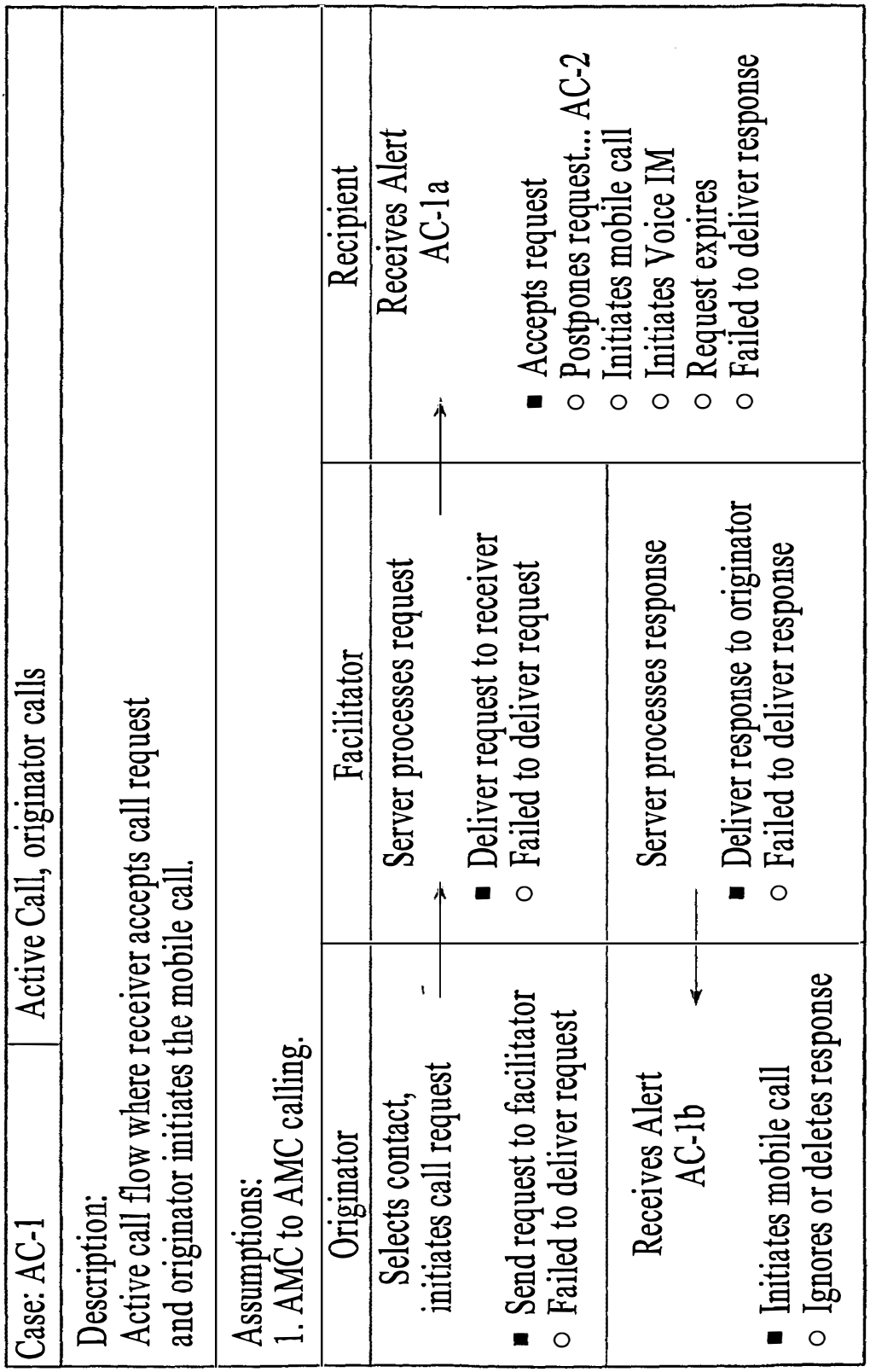

Case: AC-1
Active Call, originator calls
Description:
Active call flow where receiver accepts call request and originator initiates the mobile call.
Assumptions:
1. AMC to AMC calling.
Originator
Facilitator
Recipient
Selects contact, initiates call request
Server processes request
Receives Alert AC-1a
■ Send request to facilitator
○ Failed to deliver request
■ Deliver request to receiver
○ Failed to deliver request
■ Accepts request
○ Postpones request... AC-2
○ Initiates mobile call
○ Initiates Voice IM
○ Request expires
○ Failed to deliver response
Receives Alert AC-1b
Server processes response
■ Initiates mobile call
○ Ignores or deletes response
■ Deliver response to originator
○ Failed to deliver response

FIG.10

| | |
|---|---|
| Alert ID | AC-1a |
| Purpose | Notifies receiver of Active Call request. |
| Type | Request |
| Originator | End User, Caller |
| Originator's phone no. | Caller defined at UI.<br>Default is Caller's handset phone number |
| Display Text | N/A |
| Priority | Caller defined at UI |
| Text subject | Caller defined at UI |
| Voice subject | Caller defined at UI |
| Expiration | Caller defined at UI as [HH:MM, or wall clock time].<br>Default is 180 minutes |
| Response Options | 1. Accept call request with recipient's handset phone number (this is the default response)<br>2. Accept call request with phone number input by recipient<br>3. Initiate mobile call to originator using the value in the originator's phone number field<br>4. Postpone<br>a. 15 minutes<br>b. 60 minutes<br>c. User defined interval<br>d. User defined wall clock time<br>5. Respond with predefined text message<br>6. Initiate a Voice IM to originator<br>7. Delete |

FIG.11

| | |
|---|---|
| Alert ID | AC-1b |
| Purpose | Informs caller that recipient has accepted his Active Call request. |
| Type | Response |
| Originator | End User, Recipient |
| Display Text | N/A |
| Priority | N/A |
| Text subject | AMC generated message: "<recipient's name> accepted your call request at <time recipient responded>". |
| Voice subject | N/A |
| Expiration | Expiration time of associated AC-1a message. |
| Recipient's number | Phone number provided by AC-1a recipient |
| Response Options | 1. Initiate mobile call (using the value in the recipient's number field)<br>2. Delete |

FIG.12

| | |
|---|---|
| Alert ID | ON-1a |
| Purpose | Active call request to a non-AMC person. |
| Type | Request |
| Originator | AMC subscriber |
| Display Text | N/A |
| Priority | Caller defined at UI |
| Text subject | Caller defined at UI |
| Voice subject | N/A |
| Expiration | Caller defined at UI as [HH:MM] not to exceed 72 hours.<br>Default is 3 hours. |
| Phone number | Caller defined at UI.<br>Default is originator's mobile handset phone number. |
| Response Options | N/A |

FIG.13

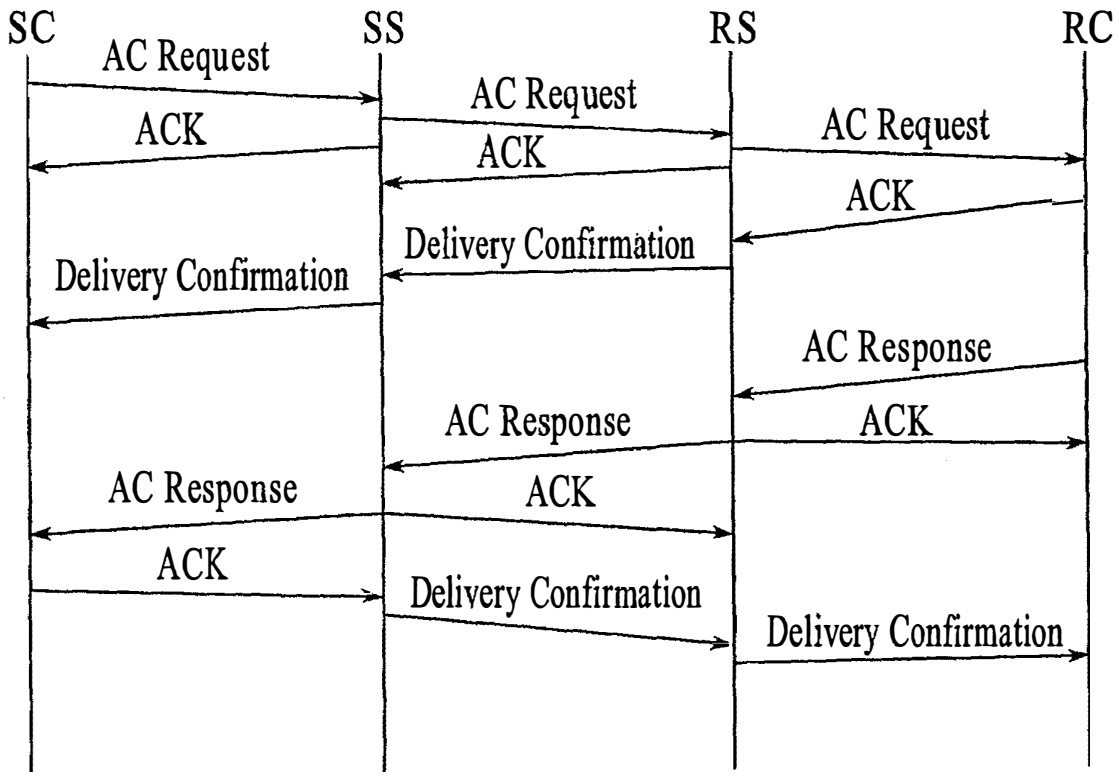

SC
SS
RS
RC
AC Request
AC Request
AC Request
ACK
ACK
ACK
Delivery Confirmation
Delivery Confirmation
AC Response
AC Response
ACK
AC Response
ACK
ACK
Delivery Confirmation
Delivery Confirmation

FIG.14

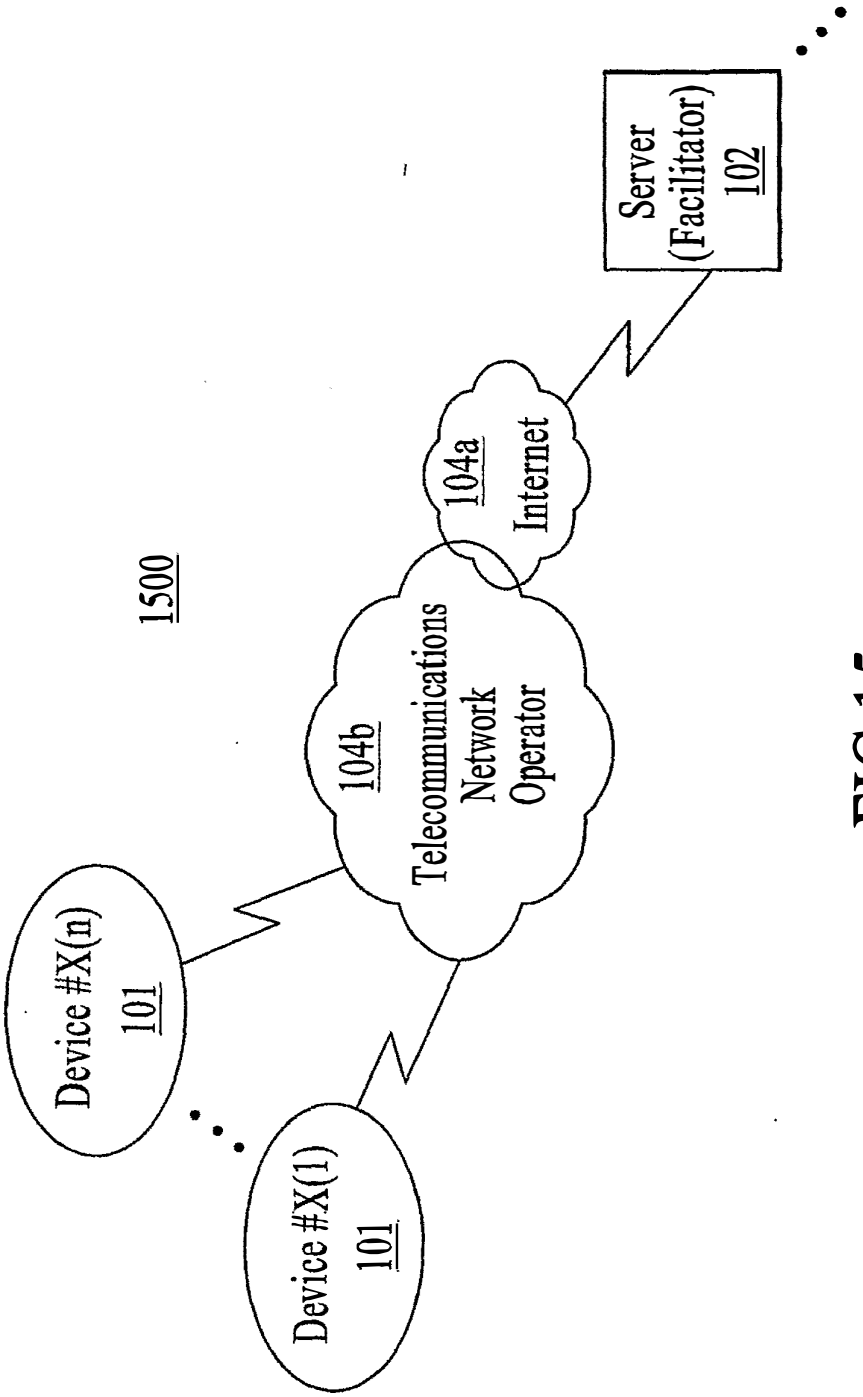

1500
Device #X(n)
101
Device #X(1)
101
104b
Telecommunications
Network
Operator
104a
Internet
Server
(Facilitator)
102

FIG.15

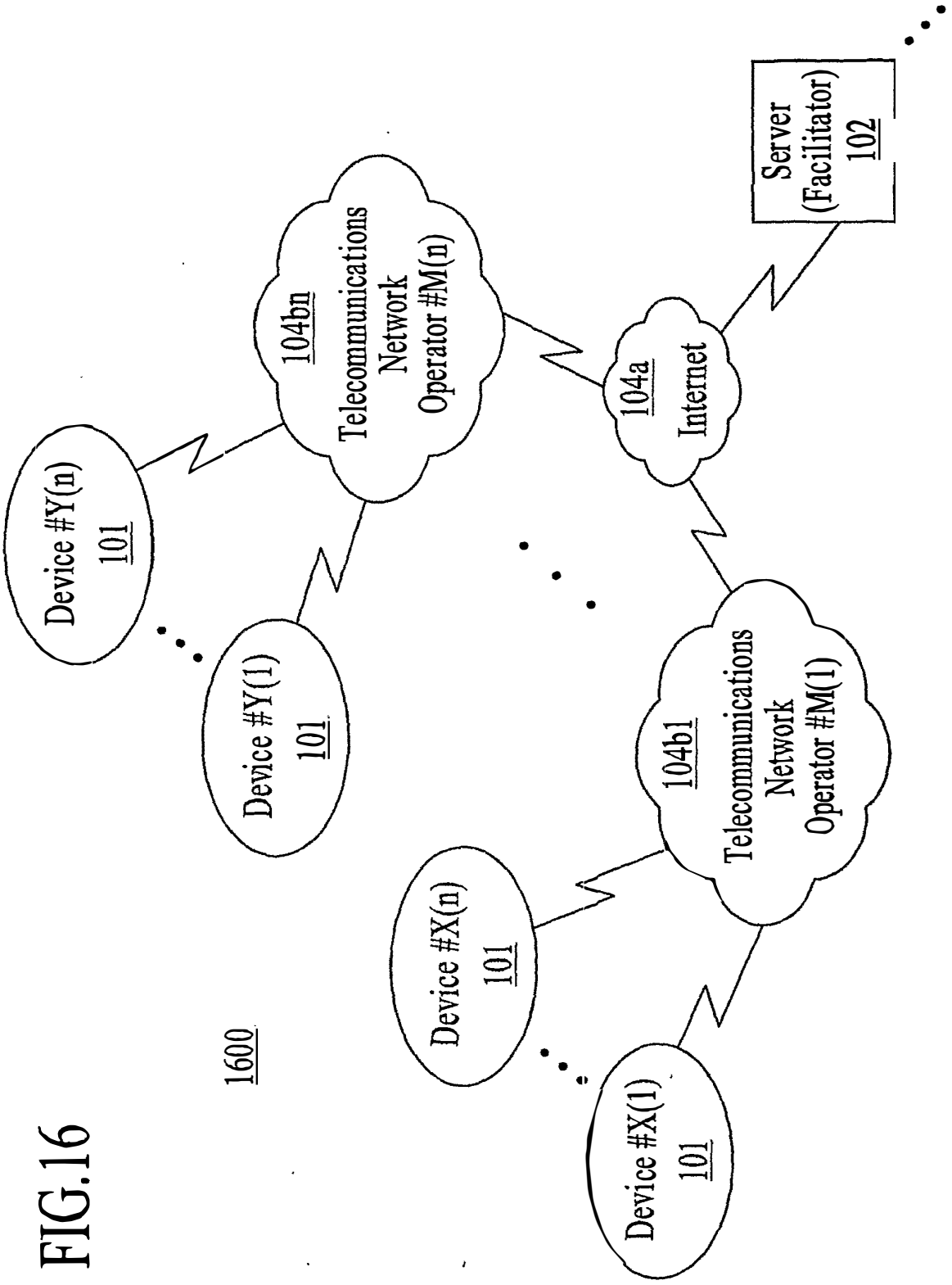

Server (Facilitator) 102
104bn Telecommunications Network Operator #M(n)
104a Internet
Device #Y(n) 101
Device #Y(1) 101
104b1 Telecommunications Network Operator #M(1)
Device #X(n) 101
Device #X(1) 101
1600

FIG.16

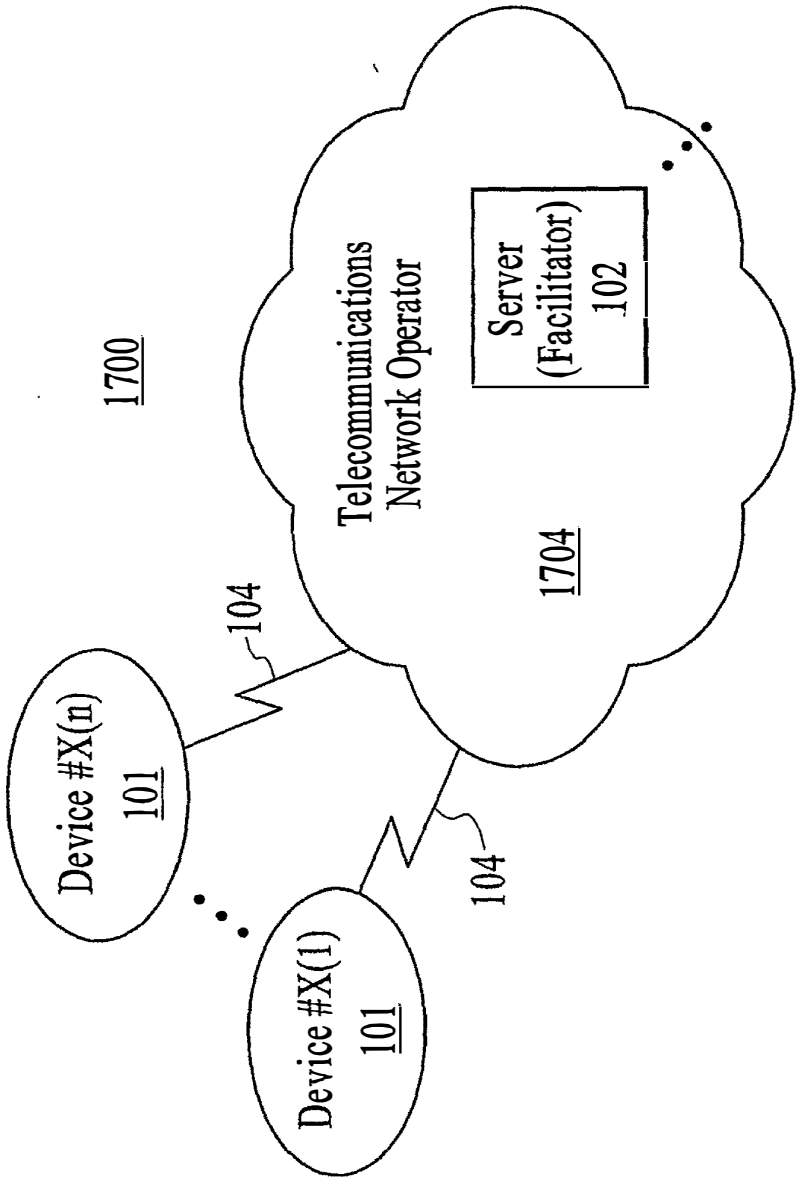

1700
Telecommunications Network Operator
1704
Server (Facilitator) 102
104
104
Device #X(n) 101
Device #X(1) 101

FIG.17

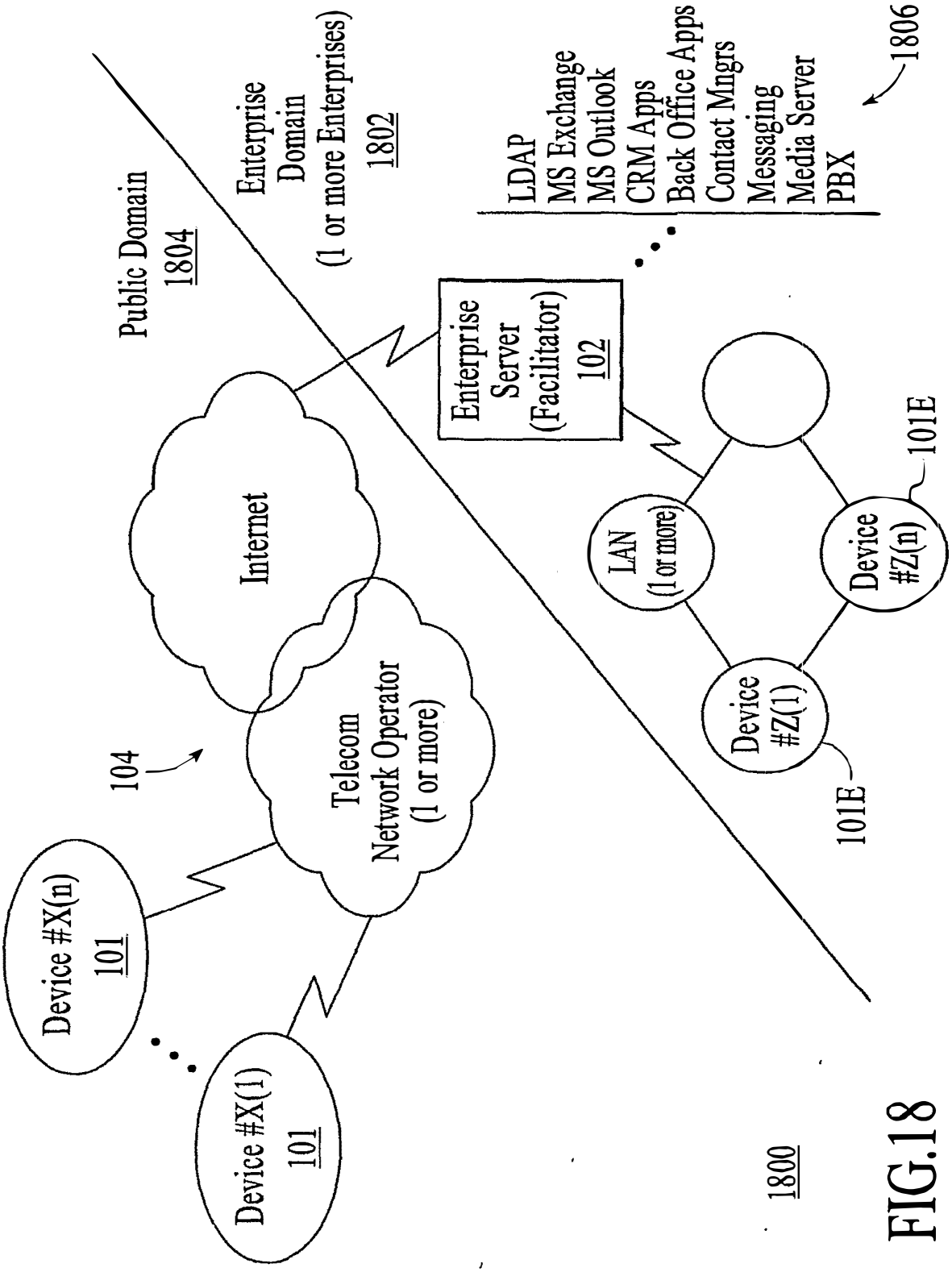

Public Domain
1804
Enterprise
Domain
(1 or more Enterprises)
1802
LDAP
MS Exchange
MS Outlook
CRM Apps
Back Office Apps
Contact Mngrs
Messaging
Media Server
PBX
1806
Enterprise
Server
(Facilitator)
102
Internet
Telecom
Network Operator
(1 or more)
104
Device #X(n)
101
Device #X(1)
101
LAN
(1 or more)
Device
#Z(1)
Device
#Z(n)
101E
101E
1800

FIG.18

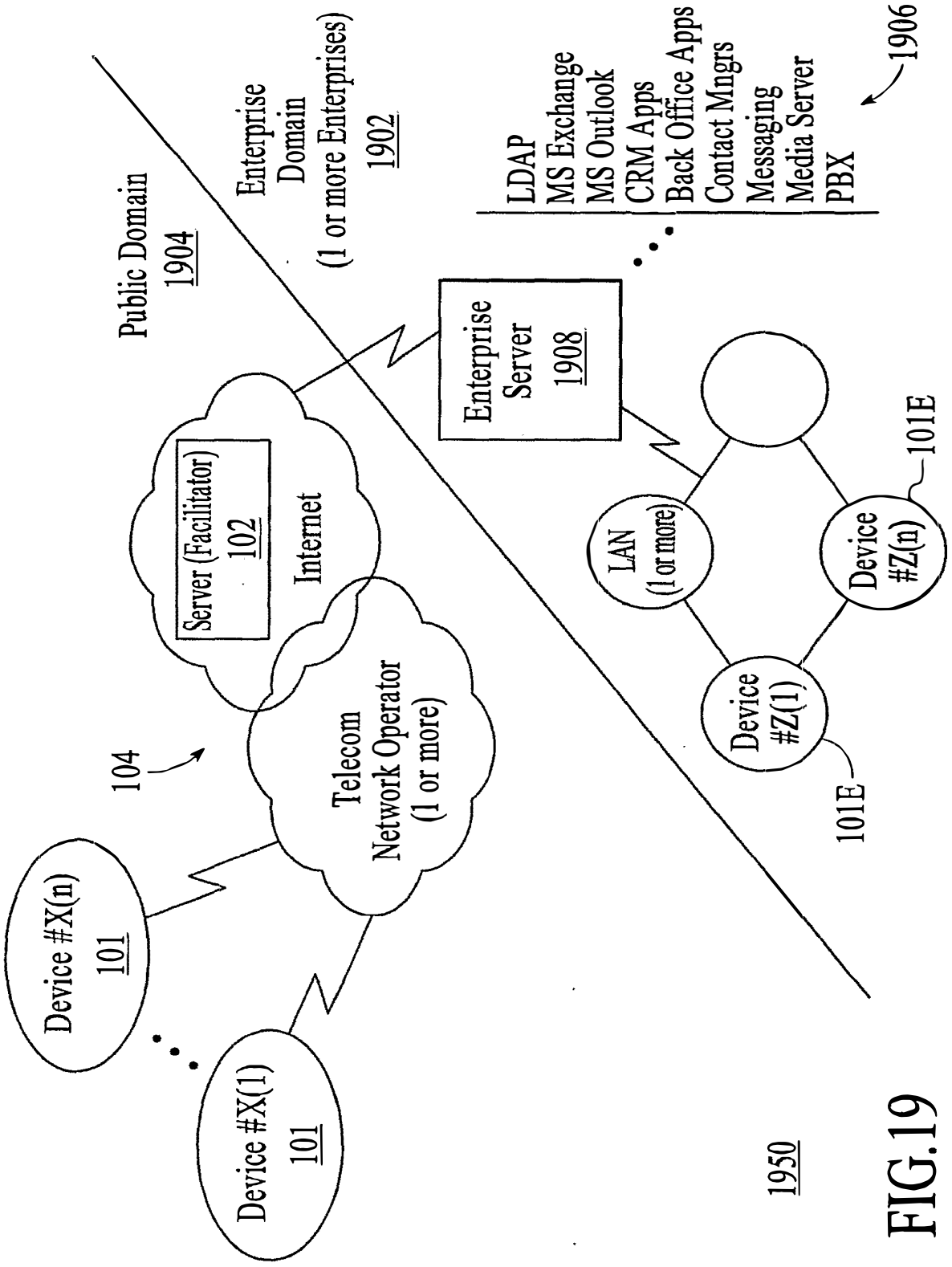

Public Domain
1904
Enterprise
Domain
(1 or more Enterprises)
1902
LDAP
MS Exchange
MS Outlook
CRM Apps
Back Office Apps
Contact Mngrs
Messaging
Media Server
PBX
1906
Enterprise
Server
1908
Server (Facilitator)
102
Internet
Telecom
Network Operator
(1 or more)
104
Device #X(n)
101
Device #X(1)
101
LAN
(1 or more)
Device
#Z(1)
Device
#Z(n)
101E
101E
1950

FIG.19

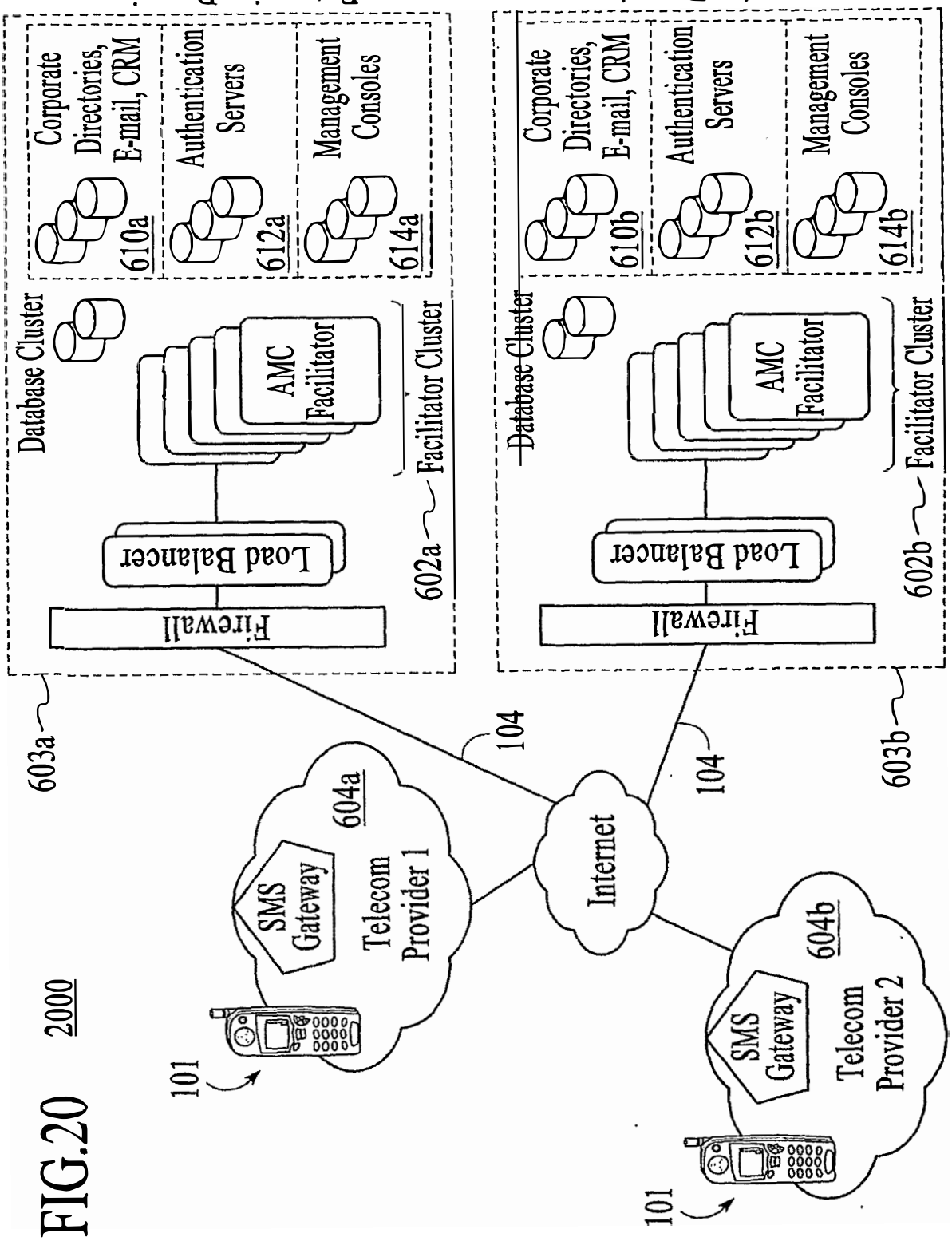

Enterprise Domain
Corporate Directories, E-mail, CRM
610a
Authentication Servers
612a
Management Consoles
614a
Database Cluster
AMC Facilitator
Load Balancer
Firewall
602a
Facilitator Cluster
603a
Enterprise Domain
Corporate Directories, E-mail, CRM
610b
Authentication Servers
612b
Management Consoles
614b
Database Cluster
AMC Facilitator
Load Balancer
Firewall
602b
Facilitator Cluster
603b
104
104
Internet
SMS Gateway
604a
Telecom Provider 1
SMS Gateway
604b
Telecom Provider 2
101
101
2000

FIG.20

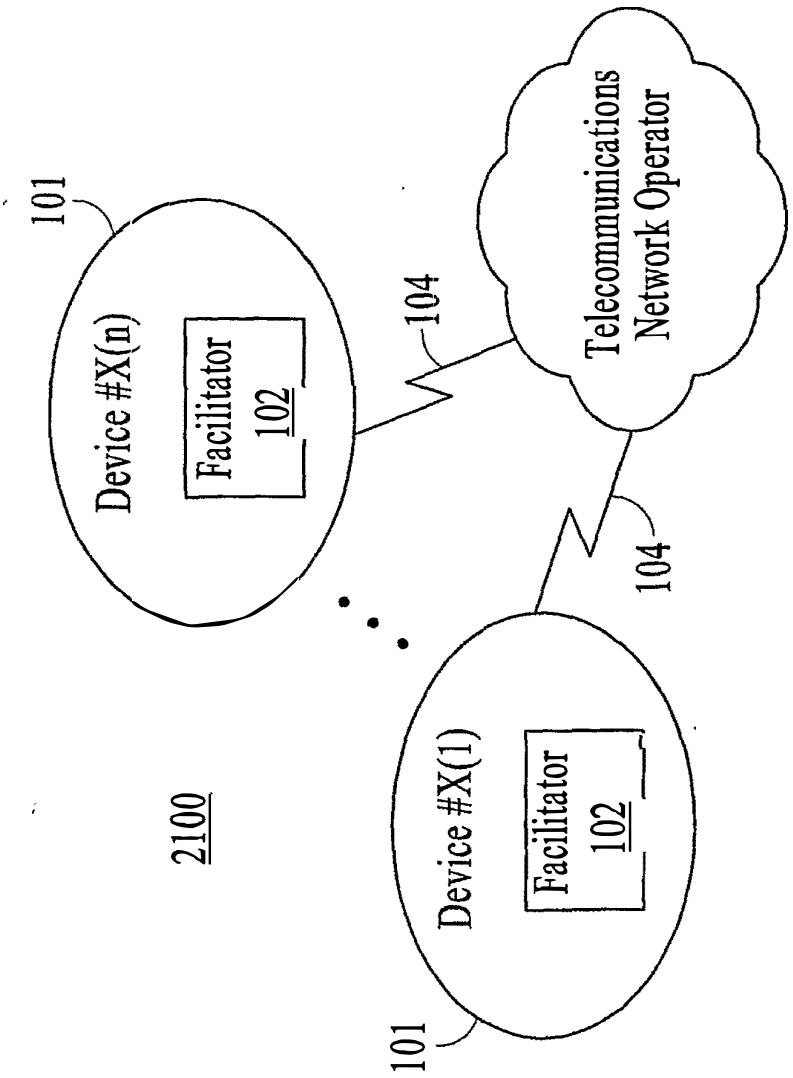

2100
101
Device #X(n)
Facilitator
102
104
Telecommunications
Network Operator
104
Device #X(1)
Facilitator
102
101

FIG.21

# Electronic Acknowledgement Receipt

| | |
|---|---|
| **EFS ID:** | 9962520 |
| **Application Number:** | 12037256 |
| **International Application Number:** | |
| **Confirmation Number:** | 3519 |
| **Title of Invention:** | TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS |
| **First Named Inventor/Applicant Name:** | Thomas E. Katis |
| **Customer Number:** | 76743 |
| **Filer:** | Steve D. Beyer/NATASHA MARCIEL |
| **Filer Authorized By:** | Steve D. Beyer |
| **Attorney Docket Number:** | REBLP003/VOX-P003 |
| **Receipt Date:** | 26-APR-2011 |
| **Filing Date:** | 26-FEB-2008 |
| **Time Stamp:** | 20:56:49 |
| **Application Type:** | Utility under 35 USC 111(a) |

## Payment information:

| | |
|---|---|
| Submitted with Payment | no |

## File Listing:

| Document Number | Document Description | File Name | File Size(Bytes)/ Message Digest | Multi Part /.zip | Pages (if appl.) |
|---|---|---|---|---|---|
| 1 | Transmittal Letter | IDS_Regular_Transmittal_5.pdf | 23796<br>7b26ac9001b9be9987a57ecee648c8bf3a08d8e6 | no | 2 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |

| | | | | | |
|---|---|---|---|---|---|
| 2 | Information Disclosure Statement (IDS) Filed (SB/08) | REBLP003_1449_for_REBL_Global_IDS_3.pdf | 30698<br>dc0a314d4c3e8840bd38ffceea5c4cc32d2ce45a | no | 2 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| This is not an USPTO supplied IDS fillable form | | | | | |
| 3 | Foreign Reference | EP000001379061A2_all_pages.pdf | 1486039<br>9ba45262b3ab65c960af8758f32a5eeb113b987f | no | 28 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 4 | Foreign Reference | EP000001715648A2_all_pages.pdf | 565930<br>34f579486322ad331db8e10c8cdc2c9b511e3804 | no | 12 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 5 | Foreign Reference | WO002006096557A2_all_pages.pdf | 1358583<br>556db682f48c65e57b41a3dd81323f2a6e7acdf1 | no | 29 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 6 | Foreign Reference | WO0020070819 29A2_all_pages.pdf | 3545409<br>d3ab861dd9c7a8e9663c8ebe72aa9d3a0eb80a9e | no | 74 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 7 | NPL Documents | 12042987_OA_11_24_10.pdf | 1294235<br>c5bb3f4f2daf2c5b786254e387ddac9cb259644c | no | 34 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 8 | NPL Documents | 12102722_12_08_10_OA.pdf | 841158<br>f621abde95df7280fa465440c0538afde537aa3d | no | 23 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 9 | NPL Documents | 12103436_12_08_10_OA.pdf | 711780<br>c286751e1c834868dcd520090f1e6501797df39f | no | 19 |
| **Warnings:** | | | | | |
| **Information:** | | | | | |
| 10 | NPL Documents | REBLP003AEP_OA_11_18_10.pdf | 117553<br>b43001cfcf8573a66288d0a3150e69afe7bf5d85 | no | 4 |

| | |
|---|---|
| **Warnings:** | |
| **Information:** | |
| **Total Files Size (in bytes):** | 9975181 |

**This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.**

**New Applications Under 35 U.S.C. 111**
**If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.**

**National Stage of an International Application under 35 U.S.C. 371**
**If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.**

**New International Application Filed with the USPTO as a Receiving Office**
**If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.**

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: Katis et al. | Attorney Docket No.: REBLP003 |
| Application No.: 12/037,256 | Examiner: Ramakrishnaiah, Melur |
| Filed: February 26, 2008 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on April 26, 2011.

Signed: /Natasha L. Marciel/
Natasha L. Marciel

# INFORMATION DISCLOSURE STATEMENT
# 37 CFR §§1.56 AND 1.97(b)

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

The reference(s) identified in the attached PTO Form 1449 (copies of non-U.S. references are attached) may be material to examination of the above-identified patent application. Applicants identify the reference(s) in compliance with their duty of disclosure pursuant to 37 CFR §§1.56 and 1.97. The Examiner is requested to make the identified reference(s) of official record in this application.

This Information Disclosure Statement is not to be construed as a representation that a search has been made, that additional information material to the examination of this application does not exist, or that the identified reference(s) indeed constitute prior art.

This Information Disclosure Statement is: (i) filed within three (3) months of the filing date of the above-referenced application, (ii) believed to be filed before the mailing date of a first Office Action on the merits, or (iii) believed to be filed before the mailing of a first Office Action after the filing of a Request for Continued Examination under §1.114. Accordingly, it is believed that no fees are due in connection with the filing of this Information Disclosure

Statement. However, if it is determined that any fees are due, the Commissioner is hereby authorized to charge such fees to Deposit Account 50-4481 (Order No. REBLP003).

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Registration No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et al. | Attorney Docket No.: REBLP003 |
| Application No.: 12/037,256 | Examiner: RAMAKRISHNAIAH, M |
| Filed: February 26, 2011 | Group: 2614 |
| Title: TELECOMMUNICATION AND MULTIMEDIAMANAGEMENT METHOD AND APPARATUS | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on July 8, 2011.

Signed: _____/Laura M. Dean/_____
Laura M. Dean

# THIRD PRELIMINARY AMENDMENT

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

Prior to examining the subject application, please amend the above-identified patent application as follows:

**Amendments to the Specification** begin on page 2 of this paper.

**Amendments to the Claims** are reflected in the listing of claims, which begin on page 3 of this paper.

**Amendments to the Drawings** begin on page 11 of this paper and include an attached replacement sheet.

**Remarks/Arguments** begin on page 12 of this paper.

**Amendments to the Specification:**

Applicant has made amendments to the specification in the attached substitute specification. Both clean and marked up copies of the specification are included, as well as the required statement under 37 CFR §1.125.The listing of claims will replace all prior versions, and listings, of claims in the application:

The listing of claims will replace all prior versions, and listings, of claims in the application:

**Listing of Claims:**

Cancel claim 1 through 43.

Please add the following new claims:

44. (New) A computer readable application embedded in a non-transitory computer readable medium and intended to run on a client communication device, the application comprising:

a network receive module configured to progressively receive from a sender media of a message transmitted over a network to the communication device;

a storage module configured to progressively store the received media of the message on the client communication device as the media is progressively received; and

a render module configured to selectively render the media of the message on the communication device, the rendering options for rendering the media of the message for the first time including both:

(i) progressively as the media of the message is progressively received over the network; and

(ii) some time after the media was received and stored by retrieving and progressively rendering the retrieved media from storage on the client communication device.

45. (New) The application of claim 44, wherein (i) further comprises progressively rendering the media of the message live as the media is created and transmitted over the network by the sender.

46. (New) The application of claim 44, wherein (i) occurs sometime after the media was created by the sender due to a delay.

47. (New) The application of claim 46, wherein the delay is caused by latency on the network.

48. (New) The application of claim 46, wherein the delay is due to the sender not connected to the network when the media of the message was created.

49. (New) The application of claim 46, wherein the delay is due to the sender designating the transmission of the message some time after the media of the message was created.

50. (New) The application of claim 44, wherein (i) further comprises progressively rendering the media of the message while the creation of the media of the message by the sender is ongoing.

51. (New) The application of claim 44, wherein (i) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

52. (New) The application of claim 44, wherein (ii) occurs after the entire message has been received and stored.

53. (New) The application of claim 44, wherein (ii) occurs while receipt and storage of the media of the message is ongoing.

54. (New) The application of claim 44, wherein (ii) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

55. (New) The application of claim 44, wherein the render module is further configured to selectively transition the progressive rendering of the media of the message from (i) to (ii).

56. (New) The application of claim 44, wherein the render module is further configured to selectively transition the progressive rendering of the media of the message from (ii) to (i).

57. (New) The application of claim 44, wherein the media of the message is time-indexed.

58. (New) The application of claim 44, further comprising a conversation module configured to associate transmitted and received messages sharing a common attribute into a conversation.

59. (New) The application of claim 58, wherein the conversation module is further configured to string together the transmitted and received messages of the conversation in time-index order.

60. (New) The application of claim 44, further comprising a transmit module configured to transmit messages originating at the client communication device.

61. (New) The application of claim 60, wherein the render module and the transmit module cooperate so that the client communication device running the application may selectively engage in synchronous communication with a remote communication device that sent the message.

62. (New) The application of claim 60, wherein the render module and the transmit module cooperate so that the client communication device running the application may selectively engage in asynchronous communication with a remote communication device that sent the message.

63. (New) The application of claim 58, wherein the media of the conversation comprises one or more of the following: voice, video, text, photos, GPS or positional data, sensor data, or any combination thereof.

64. (New) The application of claim 44, wherein the render module further includes one or more of the following rendering options: play, play faster, play slower, catch up to the head of the message, catch up to live, jump forward, jump backward and pause.

65. (New) A client communication device, comprising:

a network receive element configured to progressively receive from a sender media of a message transmitted over a network to the client communication device;

a storage element configured to progressively store the received media of the message on the client communication device as the media is progressively received; and

a render element configured to selectively render the media of the message on the client communication device, the rendering options for rendering the media of the message for the first time including both:

(i) progressively as the media of the message is progressively received over the network; and

(ii) some time after the media was received and stored by retrieving and progressively rendering the retrieved media from storage on the client communication device.

66. (New) The client communication device of claim 65, wherein (i) further comprises progressively rendering the media of the message live as the media is created and transmitted over the network by the sender.

67. (New) The client communication device of claim 65, wherein (i) occurs sometime after the media was created by the sender due to a delay.

68. (New) The client communication device of claim 67, wherein the delay is caused by latency on the network.

69. (New) The client communication device of claim 67, wherein the delay is due to the sender not connected to the network when the media of the message was created.

70. (New) The client communication of device claim 67, wherein the delay is due to the sender designating the transmission of the message some time after the media of the message was created.

71. (New) The client communication device of claim 65, wherein (i) further comprises progressively rendering the media of the message while the creation of the media of the message by the sender is ongoing.

72. (New) The client communication device of claim 65, wherein (i) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

73. (New) The client communication device of claim 65, wherein (ii) occurs after the entire message has been received and stored.

74. (New) The client communication device of claim 65, wherein (ii) occurs while receipt and storage of the media of the message is ongoing.

75. (New) The client communication device of claim 65, wherein (ii) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

76. (New) The client communication device of claim 65, wherein the render element is further configured to selectively transition the progressive rendering of the media of the message from (i) to (ii).

77. (New) The client communication of claim 65, wherein the render element is further configured to selectively transition the progressive rendering of the media of the message from (ii) to (i).

78. (New) The client communication device of claim 65, wherein the media of the message is time-indexed.

79. (New) The client communication device of claim 65, further comprising a conversation element configured to associate transmitted and received messages sharing a common attribute into a conversation.

80. (New) The client communication device of claim 79, wherein the conversation element is further configured to string together the transmitted and received messages of the conversation in time-index order.

81. (New) The client communication device of claim 65, further comprising a transmit element configured to transmit messages originating at the client communication device.

82. (New) The client communication device of claim 81, wherein the render element and the transmit element cooperate so that the client communication device may selectively engage in synchronous communication with a remote communication device that sent the message.

83. (New) The client communication device of claim 81, wherein the render element and the transmit element cooperate so that the client communication device may selectively engage in asynchronous communication with a remote communication device that sent the message.

84. (New) The client communication device of claim 79, wherein the media of the conversation comprises of one of the following: voice, video, text, photos, GPS or positional data, sensor data, or any combination thereof.

85. (New) The client communication device of claim 65, wherein the render element further includes one or more of the following rendering options: play, play faster, play slower, catch up to the head of the message, catch up to live, jump forward, jump backward and pause.

86. (New) A method, comprising:

progressively receiving at a client communication device media of a message from a sender;

progressively storing the received media of the message on the client communication device as the media is progressively received; and

selectively rendering the media of the message on the client communication device, the rendering options for rendering the media of the message for the first time including both:

(i) progressively as the media of the message is progressively received over the network; and

(ii) some time after the media was received and stored by retrieving and progressively rendering the retrieved media from storage on the client communication device.

87. (New) The method of claim 86, wherein (i) further comprises progressively rendering the media of the message live as the media is created and transmitted over the network by the sender.

88. (New) The method of claim 86, wherein (i) occurs sometime after the media was created by the sender due to a delay.

89. (New) The method of claim 88, wherein the delay is caused by latency on the network.

90. (New) The method of claim 88, wherein the delay is due to the sender not connected to the network when the media of the message was created.

91. (New) The method of device claim 88, wherein the delay is due to the sender designating the transmission of the message some time after the media of the message was created.

92. (New) The method of claim 86, wherein (i) further comprises progressively rendering the media of the message while the creation of the media of the message by the sender is ongoing.

93. (New) The method of claim 86, wherein (i) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

94. (New) The method of claim 86, wherein (ii) occurs after the entire message has been received and stored.

95. (New) The method of claim 86, wherein (ii) occurs while receipt and storage of the media of the message is ongoing.

96. (New) The method of claim 86, wherein (ii) further comprises progressively rendering the media of the message, behind the head of the message, while the creation of the media of the message by the sender is ongoing.

97. (New) The method of claim 86, further comprising selectively transitioning the progressive rendering of the media of the message from (i) to (ii).

98. (New) The method claim 86, further comprising selectively transitioning the progressive rendering of the media of the message from (ii) to (i).

99. (New) The method of claim 86, wherein the media of the message is time-indexed.

100. (New) The method of claim 86, further comprising associating transmitted and received messages sharing a common attribute into a conversation.

101. (New) The method of claim 100, further comprising stringing together the transmitted and received messages of the conversation in time-index order.

102. (New) The method of claim 86, further comprising transmitting messages originating at the client communication device.

103. (New) The method of claim 102, further comprising selectively engaging in synchronous communication with a remote communication device that sent the message.

104. (New) The method of claim 102, further comprising selectively engaging in asynchronous communication with a remote communication device that sent the message.

105. (New) The method of claim 100, wherein the media of the conversation comprises of one of the following: voice, video, text, photos, GPS or positional data, sensor data, or any combination thereof.

106. (New) The method of claim 86, wherein the selectively rendering the media of the messages further comprises using one of the following rendering options: play, play faster, play slower, catch up to the head of the message, catch up to live, jump forward, jump backward or pause.

**Amendments to the Drawings:**

Fig. 1 has been replaced with a stylized version of Fig. 1. The content is identical to the original Fig. 1. No new matter has been added.

Attachment: Replacement Sheet

## REMARKS

The Applicants have cancelled claims 1 through 43 and added new claims 44 through 106. It is respectfully requested that the new claims be examined.

Should the Examiner believe that a telephone conference would expedite the prosecution of this application, the undersigned can be reached at the telephone number set out below.

Respectfully submitted,
BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Reg. No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

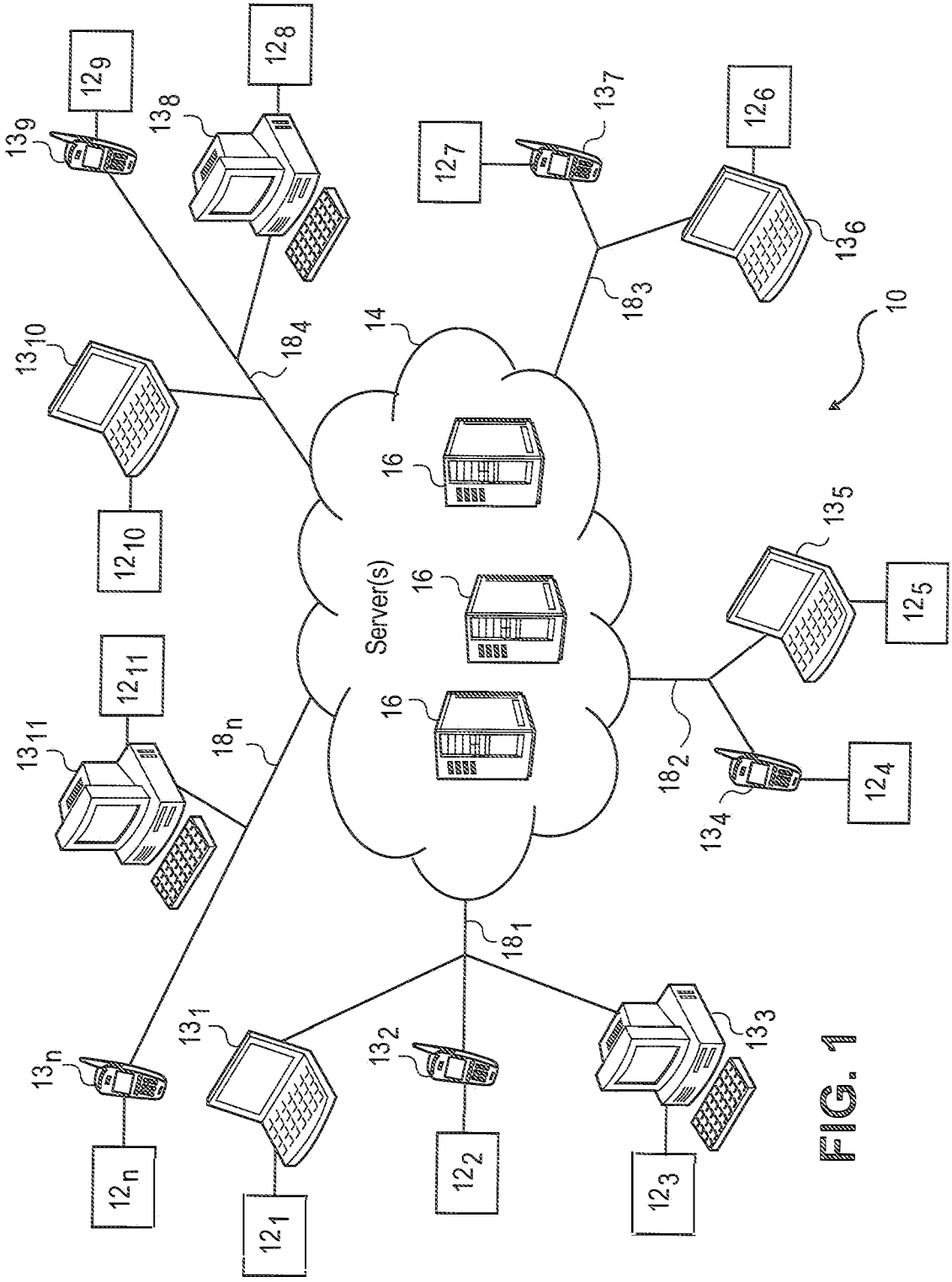

Server(s)
16
16
16
14
10
$12_1$
$12_2$
$12_3$
$12_4$
$12_5$
$12_6$
$12_7$
$12_8$
$12_9$
$12_{10}$
$12_{11}$
$12_n$
$13_1$
$13_2$
$13_3$
$13_4$
$13_5$
$13_6$
$13_7$
$13_8$
$13_9$
$13_{10}$
$13_{11}$
$13_n$
$18_1$
$18_2$
$18_3$
$18_4$
$18_n$

FIG. 1

*IN THE UNITED STATES PATENT AND TRADEMARK OFFICE*

| | |
|---|---|
| In re application of: KATIS et. al. | Attorney Docket No.: REBLP003/VOX-P003 |
| Application No.: 12/037,256 | Examiner: RAMAKRISHNAIAH, MELUR |
| Filed: February 26, 2008 | |
| Title: TELECOMMUNICATION AND MULTIMEDIA MANGEMENT METHOD AND APPARATUS | Group: 2614 |
| | Confirmation No.: 3519 |

CERTIFICATE OF EFS-WEB TRANSMISSION
I hereby certify that this correspondence is being transmitted electronically through EFS-WEB to the Commissioner for Patents, P.O. Box 1450 Alexandria, VA 22313-1450 on July 8, 2011.

Signed: _____/Laura M. Dean/_____
Laura M. Dean

# SUBMISSION OF SUBSTITUTE SPECIFICATION

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

Enclosed is a Substitute Specification for the originally filed specification in this application. Also enclosed is a marked-up copy of the Substitute Specification showing the matter being added to and the matter being deleted from the specification.

Provided below is a statement as required by 37 CFR §1.125, that the Substitute Specification transmitted herewith includes no new matter.

## STATEMENT THAT THE SUBSTITUTE SPECIFICATION INCLUDES NO NEW MATTER (37 CFR §1.125)

I hereby state that the accompanying Substitute Specification includes no new matter over that contained in the above-identified application originally filed.

It is submitted, therefor, that the Substitute Specification should be accepted by the Examiner and entered into the file. If there are any questions regarding this matter, the Examiner is respectfully requested to telephone the undersigned.

Respectfully submitted,

BEYER LAW GROUP LLP

/Steve D Beyer/
Steve D Beyer
Reg. No. 31,234

P.O. Box 1687
Cupertino, CA 95015-1687
408-255-8001

# SUBSTITUTE SPECIFICATION

# MARKED UP COPY

# TELECOMMUNICATION AND MULTIMEDIA MANAGEMENT METHOD AND APPARATUS

## CROSS-REFERENCE TO RELATED APPLICATIONS

**[0001]** This application is a continuation of U.S. Application 12/028,400, filed on February 8, 2008 entitled "Telecommunication and Multimedia Management Method and Apparatus", which claims the benefit of U.S. Provisional Patent Application No. 60/937,552, filed June 28, 2007 entitled "Re-thinking Voice Communication" and U.S. Provisional Patent Application No. 60/999,619, filed on October 19, 2007 entitled "Telecommunication and Multimedia Management System and Method", all of which are incorporated herein by reference for all purposes.

## BACKGROUND

Field of the Invention.

**[0002]** This invention pertains to telecommunications, and more particularly, to a telecommunication and multimedia management method and apparatus that enables users to review the messages of conversations in either a live mode or a time-shifted mode and to transition the conversation back and forth between the two modes, participate in multiple conversations and to archive the messages of conversations for later review or processing.

Description of Related Art

**[0003]** The current state of voice communications suffers from inertia. In spite of automated switching, high bandwidth networks and technologies such as satellites, fiber optics, Voice over IP (VoIP), wireless and cellular networks, there has been little change in how people use telephones. One is still required to pick up the phone, dial another party, wait for a connection to be made, and then engage in a full-duplex, synchronous conversation with the dialed party. If the recipient does not answer, no connection is made, and the conversation does not take place.

**[0004]** At best, a one-way asynchronous voice message may be left if the recipient has voice mail. The process of delivering the voice mail, however, is burdensome and time consuming. The caller is required to wait for the phone on the other end to stop ringing, transition into the voice mail system, listen to a voice message greeting, and then leave the message. Current voice mail systems are also inconvenient for the recipient. The recipient has to dial a code to access their voice mail, navigate through a series of prompts, listen to any earlier received voice messages in the queue, and then finally listen to the message of the sender.

**[0005]** Another drawback with typical voice mail systems is the inability to organize or permanently archive voice messages. With some voice mail systems, a user may save a message, but it is automatically deleted after a predetermined period of time and lost forever.

**[0006]** Yet another problem with current voice mail systems is that a connection must be made between the caller and the voice mail system before a message can be left. If no connection is made, there is no way for the caller to leave a message.

**[0007]** Current telephone systems are based on relatively simplistic usage patterns: real-time live calls or disjointed voice mail messages, which are typically deleted as they are heard. These forms of voice communications do not capture the real power that can be achieved with voice communication or take advantage of the advances of network speed and bandwidth that is now available. Also, if the phone network is down, or is inaccessible, (e.g., a cell phone user is in an area of no coverage or the phone lines are down due to bad weather), no communication can take place.

**[0008]** In general, telephone based communications have not kept pace with the advances in text-based communications. Instant messaging, emailing, faxing, chat groups, and the ability to archive text messages, are all commonplace with text based communications. Other than voice mail, there are few existing tools available to manage and/or archive voice messages. In comparison, the tools currently available to manage telephone communications are primitive compared to text communications.

**[0009]** The corporate environment provides just one example of the weakness in current voice communication tools. There is currently no integrated way to manage voice communications across an organization as a corporate asset. Employees generally do not record or persistently store their phone conversations. Most business related

voice communication assets are gone as quickly as the words are spoken, with no way to manage or store the content of those conversations in any manageable form.

**[0010]** As an illustrative example, consider a sales executive at a company. During the course of a busy day, the executive may make a number of calls, and close several sales, with customers over the phone. Without the ability to organize, store, and later retrieve these conversations, there is no way for the executive to resolve potential issues that may arise, such as recalling the terms of one deal versus another, or challenging a customer who disputes the terms of a previously agreed upon sale. If this executive had the ability to easily retrieve and review conversations, these types of issues could be easily and favorably resolved.

**[0011]** Current tactical radio systems, such as those used by the military, fire, police, paramedics, rescue teams, and first responders, also suffer from a number of deficiencies. Most tactical radio communication must occur through a "live" radio connection between the sender of a message and a recipient. If there is no radio connection between the two parties, there can be no communication. Urgent messages cannot be sent if either the sender or the receiver does not have access to their radio, or a radio circuit connection is not established. Tactical communications are therefore plagued with several basic problems. There is no way (i) to guarantee the delivery of messages, (ii) for a recipient to go back and listen to a message that was not heard in real time; (iii) to control the granularity of the participants in a conversation; (iv) for the system to cope when there is a lack of signal integrity for a live conversation. If a message is not heard live, it is missed. There are no tools for either the sender or a recipient to manage, prioritize, archive and later retrieve (i.e. time-shift) the messages of a conversation that were previously sent.

**[0012]** Yet another drawback with tactical radio communication systems is that only one message can be sent at a time per channel. Consider an example of a large building fire, where multiple teams of fire fighters, police and paramedics are simultaneously rescuing victims trapped in the building, fighting the fire, providing medical aid to victims, and controlling bystanders. If each of the teams is using the same channel, communications may become crowded and chaotic. Transmissions get "stepped on" when more than one person is transmitting at the same time. Also there is no way to differentiate between high and low priority messages. A team inside the burning building fighting the fire or rescuing trapped victims should have a higher priority over

other teams, such as those controlling bystanders. If high priority messages are stepped on by lower priority messages, it could not only hamper important communications, but could endanger the lives of the fire fighters and victims in the building.

**[0013]** One possible solution to the lack of ability to prioritize messages is to use multiple channels, where each team is assigned a different channel. This solution, however, creates its own set of problems. How does the fire chief determine which channel to listen too at any point in time? How do multiple teams communicate with one another if they are all on different channels? If one team calls for urgent help, how are other teams to know if they are listening to other channels? While multiple channels can alleviate some issues, it can also cause confusion, creating more problems than if a single channel is used.

**[0014]** The lack of management tools that effectively prioritize messages, that allow multiple conversations to take place at the same time, that enable the time-shifting of messages to guarantee delivery, or that support archiving and storing conversations for later retrieval and review, all contribute to the problems associated with tactical radios. In first responder situations, such as with the military, police, and fire, effective communication tools can literally mean the difference between life and death, or the success or failure of a mission. The above burning building example is useful in illustrating just some of the issues with current tactical radio communications. Similar problems exist with the military, police, first responders and others who use tactical communications.

**[0015]** With packet-based networks, commonly used protocols include Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). UDP offers the advantage of fast delivery of data, but at the expense of completeness. Packets may be dropped in transit and not available when attempting to render the data as soon as possible at the destination. In spite of the shortcomings, UDP is the standard for Voice over Internet Protocol (VoIP) transmissions due to its speed attributes. On the other hand TCP does guarantee the delivery of perfect (i.e., an exact copy of the transmitted data) data, but at the expense of latency. All packets are delivered, regardless of how long it takes. This delay makes TCP impractical for use with "live" phone calls. Currently there are no known protocols that offer the performance advantages of both TCP and UDP, where "good enough" media can be transmitted for rendering as soon as possible, with the eventual delivery of a perfect copy of the media. Also there is no protocol that

determines how much information should be sent over the network based on the presence of recipients on the network and their intentions to render the data either live or in a time-shifted mode. In addition, other factors commonly considered, such as network latency, network degradation, packet loss, packet damage, and general bandwidth conditions, are used in determining how much data to transmit. Prior art systems, however, do not consider the presence and intentions of recipients. As a result, the default assumption is that the data is rendered by the recipient in real time. When a recipient is not going to render data immediately, these prior art systems unnecessarily use bandwidth when it is not needed, degrading the overall performance of the network.

**[0016]** For the reasons recited above, telephone, voicemail and tactical voice communications systems are inadequate. An improved voice and media communication and management system and method, and improvements in delivering voice and other media over packet-based networks, is therefore needed.

## SUMMARY OF THE INVENTION

~~[0017] The present invention is directed to a communication method for communicating on a first communication device. The method includes receiving at the first communication device media of a conversation with one or more participants over a network and storing on the first communication device the received media as the media is received over the network. The received media of the conversation may be rendered in either a near-real-time mode or a time-shifted mode. The rendering of the media may be transitioned on the first communication device from the time-shifted mode to the near-real-time mode by retrieving the stored media, rendering the retrieved media from storage at an increased rendering rate, ascertaining a catch-up point when the rendering of the retrieved media at the increased rate has caught up and is substantially concurrent with the media of the conversation as it is being received over the network and rendering the media of the conversation in the near-real-time mode as it is being received over the network after the catch-up point.~~

[0018] The above-described problems are solved by a communication application, method and apparatus that supports new modes of engaging in conversations and/or managing multiple conversations using a variety of media types, such as voice, video, text, location, sensor information, and other data. Users can engage in conversations by sending messages to designated recipients. Depending on preferences and priorities, the recipient(s) might participate in the conversation in real time, or they might simply be notified that the message is ready for retrieval. In the latter case, the recipient participates in the conversation in a time-shifted mode by reviewing and replying to the recorded message at their convenience.

[0019] Users are empowered to conduct communications in either: (i) a near-synchronous or "live" conversation, providing a user experience similar to a standard full duplex phone call; or (ii) in a series of back and forth time-delayed transmissions (i.e., time-shifted mode). Further, users engaged in a conversation can seamlessly transition from the live mode to the time-shifted mode and back again. This attribute also makes it possible for users to engage in multiple conversations, at the same time, by prioritizing and shifting between the two modes for each conversation. Two individuals using the system can therefore send

**recorded voice messages back and forth to each other and review the messages when convenient, or the messages can be sent at a rate where they essentially merge into a live, synchronous voice conversation. This new form of communication, for the purposes of the present application, is referred to as "Voxing"**

**[0020]** **When you "Vox" someone, the conversation consists of a series of discrete recorded messages, which are recorded in a number of locations, which may include the encoding device of the sender, (e.g. a phone or computer), servers on multiple transmission hops across the network, and the receiver's rendering device. Unlike a standard phone call or voice mail, the system provides the following features and advantages: (i) the conversation can transition between live and time-shifted or vice versa; (ii) the discrete messages of the conversation are semantically threaded together and archived; (iii) since the messages are recorded and are available for later retrieval, attention can be temporarily diverted from the conversation and then the conversation can be later reviewed when convenient; (iv) the conversation can be paused for seconds, minutes, hours, or even days, and can be picked up again where left off; (v) one can rejoin a conversation in progress and rapidly review missed messages and catch up to the current message (i.e., the live message); (vi) no dedicated circuit is needed for the conversation to take place, as required with conventional phone calls; and (vii) lastly, to initiate a conversation, one can simply begin transmitting to an individual or a group. If the person or persons on the other end notice that they are receiving a message, they have the option of reviewing and conducting a conversation in real time, or reviewing at a later time of their choice.**

**[0021]** **The communication media management system also supports new modes of optimizing the transmission of data over a network. The system actively manages the delivery of payloads to a recipient engaged in a conversation in real time when network conditions are less than ideal. For example when network conditions are poor, the system intentionally reduces the quality of the data for transmission to the point where it is "good enough" to be rendered upon receipt by the recipient, allowing the real time participation of the conversation. The system also guarantees the eventual delivery of an "exact" copy of the messages over time. The system and method therefore provides the advantages of both speed and accuracy.**

**The utilization of network bandwidth is optimized by making tradeoffs between timeliness and media quality, using the presence and intentions of whether or not recipient(s) intend to review to the message immediately in real time, as well as measures of network latency, network degradation, packet loss or damage, and/or current bandwidth conditions.**

**[0022] It should be noted that the messages of conversations may contain voice only or voice, video and other data, such as sensor information. When the messages are reviewed, they are listened to or visually reviewed, or a combination thereof, depending on the type of media contained in the messages. Although as of the filing of the present application, most conversations are voice only, it is intended that the communication system and method described herein broadly includes conversations including multiple media types, such as voice and video for example.**

## BRIEF DESCRIPTION OF THE DRAWINGS

**[0023]** The invention may best be understood by reference to the following description taken in conjunction with the accompanying drawings, which illustrate specific embodiments of the invention.

**[0024]** Figure 1 is a diagram of the architecture of the communication and media management system of the invention.

**[0025]** Figures 2A and 2B illustrate a block diagram of a Client running on a Device in the communication and management system of the invention.

**[0026]** Figure 3 is a block diagram of a Server used in the communication and media management system of the invention.

**[0027]** Figures 4A through 4D illustrate various embodiments of data payloads used in the communication and management system of the invention.

**[0028]** Figure 5 is a diagram illustrating data being transmitted over a shared IP network in accordance with the invention.

**[0029]** Figure 6 is a diagram illustrating data being transmitted over a circuit based network in accordance with the invention.

**[0030]** Figure 7 is a diagram illustrating data being transmitted across both a cellular network and the Internet in accordance with the invention.

**[0031]** Figures 8A through 8F are a series of flow diagrams illustrating a store and stream function of the communication and management system of the invention.

**[0032]** Figures 9A through 9C are flow diagrams illustrating the operation of a Payload Quality Manager (PQM) and Figures 9D through 9F are flow diagrams illustrating the Data Quality manager (DQM), both used by the Clients and Servers of the invention.

**[0033]** Figure 10 is an exemplary device having a graphical user interface that may be used with the system of the invention.

**[0034]** Figures 11A through 11F are diagrams illustrating multiple conversation management (MCMS) features of the invention.

**[0035]** Figures 12A through 12C are diagrams illustrating the multiple conversation management system – consecutive (MCMS-C) features of the invention.

**[0036]** Figures 13A through 13D illustrate a series of diagrams detailing the operation of the invention.

**[0037]** Figures 14A and 14B are block diagrams that illustrate the hardware used for running the Client and Server applications of the invention.

**[0038]** It should be noted that like reference numbers refer to like elements in the figures.

## DETAILED DESCRIPTION OF SPECIFIC EMBODIMENTS

**[0039]** The invention will now be described in detail with reference to various embodiments thereof as illustrated in the accompanying drawings. In the following description, specific details are set forth in order to provide a thorough understanding of the invention. It will be apparent, however, to one skilled in the art, that the invention may be practiced without using some of the implementation details set forth herein. It should also be understood that well known operations have not been described in detail in order to not unnecessarily obscure the invention.

### A. Functional Overview

**[0040] The communication media management method and system supports new modes of engaging in voice conversations and/or managing multiple simultaneous conversations using a variety of media types, such as voice, video, text, location, sensor information, and other data. Users can engage in conversations by sending voice messages to designated recipients. Depending on preferences and priorities, the recipient(s) might participate in the conversation in real time, or they might simply be notified that the message is ready for retrieval. In the latter case, the recipient participates in the conversation in a time-shifted mode by reviewing and replying to the recorded message at their convenience.**

**[0041] Users are empowered to conduct communications in either: (i) a near-synchronous or "live" conversation, providing a user experience similar to a standard full duplex phone call; or (ii) in a series of back and forth time-delayed transmissions (i.e., time-shifted mode). Further, users engaged in a conversation can seamlessly transition from the live mode to the time-shifted mode and back again. This attribute also makes it possible for users to engage in multiple conversations, at the same time, by prioritizing and shifting between the two modes for each conversation. Two individuals using the system can therefore send recorded voice messages back and forth to each other and review the messages when convenient, or the messages can be sent at a rate where they essentially merge into a live, synchronous voice conversation. This new form of communication, for the purposes of the present application, is referred to as "Voxing"**

**[0042] When you "Vox" someone, the conversation consists of a series of discrete recorded messages, which are recorded in a number of locations, which may include the encoding device of the sender, (e.g. a phone or computer), servers on multiple transmission hops across the network, and the receiver's rendering device. Unlike a standard phone call or voice mail, the system provides the following features and advantages: (i) the conversation can transition between live and time-shifted or vice versa; (ii) the discrete messages of the conversation are semantically threaded together and archived; (iii) since the messages are recorded and are available for later retrieval, attention can be temporarily diverted from the conversation and then the conversation can be later reviewed when convenient; (iv) the conversation can be paused for seconds, minutes, hours, or even days, and can be picked up again where left off; (v) one can rejoin a conversation in progress and rapidly review missed messages and catch up to the current message (i.e., the live message); (vi) no dedicated circuit is needed for the conversation to take place, as required with conventional phone calls; and (vii) lastly, to initiate a conversation, one can simply begin transmitting to an individual or a group. If the person or persons on the other end notice that they are receiving a message, they have the option of reviewing and conducting a conversation in real time, or reviewing at a later time of their choice.**

**[0043] The communication media management system also supports new modes of optimizing the transmission of data over a network. The system actively manages the delivery of payloads to a recipient engaged in a conversation in real time when network conditions are less than ideal. For example when network conditions are poor, the system intentionally reduces the quality of the data for transmission to the point where it is "good enough" to be rendered upon receipt by the recipient, allowing the real time participation of the conversation. The system also guarantees the eventual delivery of an "exact" copy of the messages over time. The system and method therefore provides the advantages of both speed and accuracy. The utilization of network bandwidth is optimized by making tradeoffs between timeliness and media quality, using the presence and intentions of whether or not recipient(s) intend to review to the message immediately in real time, as well as measures of network latency, network degradation, packet loss or damage, and/or current bandwidth conditions.**

~~**[0044] It should be noted that the messages of conversations may contain voice only or voice, video and other data, such as sensor information. When the messages are reviewed, they are listened to or visually reviewed, or a combination thereof, depending on the type of media contained in the messages. Although as of the filing of the present application, most conversations are voice only, it is intended that the communication system and method deseribed herein broadly includes conversations including multiple media types, such as voice and video for example.**~~

**[0045]** An improved voice and other media communication **application, method** and ~~**management system and method**~~ **apparatus** is disclosed which provides one or more of the following features and functions:

i. enabling users to participate in multiple conversation types, including live phone calls, conference calls, voice messaging, consecutive or simultaneous communications;

ii. enabling users to review the messages of conversations in either a live mode or a time-shifted mode (voice messaging);

iii. enabling users to seamlessly transition a conversation between a synchronous "live" mode and a time shifted mode;

iv. enabling users to participate in conversations without waiting for a connection to be established with another participant or the network. This attribute allows users to begin conversations, participate in conversations, and review previously received time-shifted messages of conversations even when there is no network available, when the network is of poor quality, or other participants are unavailable;

v. enabling the system to save media payload data at the sender and, after network transmission, saving the media payload data at all receivers;

vi. enabling the system to organize messages by threading them sequentially into semantically meaningful conversations in which each message can be identified and tied to a given participant in a given conversation;

vii. enabling users to manage each conversation with a set of user controlled functions, such as reviewing "live", pausing or time shifting the conversation until it is convenient to review, replaying in a variety of modes (e.g., playing faster, catching up to live, jump to the head of the conversation) and methods for managing conversations (archiving, tagging, searching, and retrieving from archives);

viii. enabling the system to manage and share presence data with all conversation participants, including online status, intentions with respect to reviewing any given message in either the live or time-shifted mode, current attention to messages, rendering methods, and network conditions between the sender and receiver;

ix. enabling users to manage multiple conversations at the same time, where either (a) one conversation is current and all others are paused; (b) multiple conversations are rendered consecutively, such as but not limited to tactical communications; or (c) multiple conversations are active and simultaneously rendered, such as in a stock exchange or trading floor environment.

x. enabling users to store all conversations, and if desired, persistently archive them in a tangible medium, providing an asset that can be organized indexed, searched, transcribed, translated and/or reviewed as needed;

xi. enabling the system to provide real time call functionality using a best-efforts mode of message delivery at a rate "good enough" for rendering as soon as possible (similar to UDP), and the guaranteed eventual delivery of exact copies of the messages as transmitted by requesting retransmission of any missing or defective data from the originally saved perfect copy (similar to TCP); and

xii. enabling the system to optimize the utilization of network bandwidth by making tradeoffs between timeliness and media quality, using the presence and intentions of the recipient(s) (i.e., to either review the media in real-time or in a time-shifted mode), as well as measures of network latency, network degradation, packet loss or damage, and/or current bandwidth conditions.

**[0046]** In various embodiments, some or all of the numerous features and functions listed above may be implemented. It should be understood, however, that different embodiments of the invention need not incorporate all of the above listed features and functions.

## B. Glossary

**[0047]** Prior to explaining the details of the invention, it is useful to define some of the terms and acronyms used throughout the written description. This glossary of terms is organized into groups of System Components, Media, Media Management, People and Conversation Management.

### B.1. System Components

**[0048] Client**: A Client is the user application in the communication system, which includes a user interface, persistent data storage, and "Voxing" functionality. Users interact with the Client application, and the Client application manages all communications (messages and signals) and payload (Media) transfers that are transmitted or received over a network. The Client supports encoding of media (e.g., the capturing of voice, video, or other data content) and the rendering of media and supports security, encryption and authentication as well as the optimization of the transmission of data across the network. A Client may be used by one or multiple Users (i.e., multi-tenant).

**[0049] Device**: A physical device that runs the Client application. A User may be actively logged into a single Device or multiple Devices at any given point of time. In various embodiments, a Device may be a general-purpose computer, a portable computing device, a programmable phone, a programmable radio, or any other programmable communication device.

**[0050] Servers**: A computer node on the communication network. Servers are responsible for routing Messages sent back and forth between Users over the network and the persistent storage and archiving of Media payloads. Servers provide routing, transcoding, security, encryption and authentication and the optimization of the transmission of data across the network.

### B.2. Media

**[0051] Message**: An individual unit of communication from one User to another. Each Message consists of some sort of Media, such as voice or video. Each Message is assigned certain attributes, including: (i) the User sending the message; (ii) the Conversation it belongs to; (iii) an optional or user created Importance Tag; (iv) a time stamp; and (v) the Media payload.

**[0052] Media**: Audio, video, text, position, sensor readings such as temperature, or other data.

**[0053] Conversation**: A thread of Messages (identified, persistently stored, grouped, and prioritized) between two or more Users on their Devices. Users generally participate in a Conversation using their Devices by either Reviewing Messages in real time or in a time-shifted mode, or creating and sending Messages of a Conversation as

desired. When new Messages are created, they either define a new Conversation, or they are added to an existing Conversation.

**[0054] Head of a Conversation**: The most recent Message of a conversation that has been encoded by the most recent speaker. It is where a User is positioned in a Conversation when reviewing "live" or where one jumps to if the "Jump To Live" feature is used.

**[0055] Multiple Conversation Management System or MCMS**: An application that runs as part of a Client application, which enables a User to engage in multiple Conversations using a variety of Media types. With the MCMS application, a User selects one Conversation among the multiple Conversations as current, where only the Messages of current conversation are rendered. For the selected current Conversation, the User may transition from a series of back and forth Messages in time-shifted mode to near-synchronous "live" mode, similar to a standard telephone conversation, and back again. The Messages of the non-selected Conversations are in a paused state. Messages associated with the non-selected Conversion will accumulate if others are still participating in those Conversations. The User may selectively transition the current Conversation among the multiple Conversations and Review the accumulated Messages of the selected current Conversation.

**[0056] Multiple Conversation Management System-Consecutive or MCMS-C**: Similar to MCMS, with the added feature of rendering and enabling Users to manage and participate in multiple Conversations consecutively through a hierarchical system of Priorities and time-shifting, which are automatically managed by the system. The MCMS-C application allows the Messages of consecutive Conversations to be rendered in a prioritized order, as opposed to MCMS where only the Messages of the currently selected Conversation are rendered. MCMS-C is particularly applicable in situations where it is important that the Messages of the consecutive Conversations are rendered, in the prioritized order, and/or the receipt of all Messages, even those belonging to lower priority Conversations, is more important than receiving the Messages in real-time. Examples of situations where MCMS-C may be suitable include, but are not limited to, hospitals, taxi fleet management, or tactical communications.

**[0057] Multiple Conversation Management System-Simultaneous or MCMS-S**: Similar to MCMS, with the added feature of enabling With MCMS-S, multiple

Conversations are selected for simultaneous rendering, as opposed to MCMS where the Messages of only the selected current Conversation are rendered. The MCMS-S application is particularly applicable in situations where a User is listening to multiple Conversations at the same time, such as a trader listening to multiple brokers on different exchanges and periodically sending trading requests to one or multiple of them simultaneously. MCMS-S may also be suitable for tactical communications as well.

**[0058] Priority**: The mechanism through which the system determines which Message to render next when a User is participating in MCMS-C. Priority is automatically managed by the system. A User can set default Priorities, or a predetermined set of system Priorities may be used. In the event of a conflict, where more than one Message is ready to be rendered at the same time, the system resolves the conflict at least partly based on Priority, to determine what Message to render immediately and what Message to time shift.

**[0059] Tags**: a set of attributes a User or the system may assign to a Conversation or a message, such as a topic (a company name), a directive ("action items"), a indicator ("conversation summary"), or any other label by which one might want to search or organize the data.

**[0060] Importance Tags**: A special Message attribute that enables a sender to specify when a Message is to be rendered, regardless of other Priority settings. An "urgent" Importance tag will override other Priorities for example. This feature is critical for tactical systems, though any system can be configured to use or disable this feature.

**[0061] Packet**: Any unit of binary data capable of being routed through a computer network. Each packet consists of header (meta data) and payload (media data). Includes standard packet protocols such as, but not limited to, Internet Protocol (IP), EvDO, UMTS or any other packet-based network, either radio, fiber optic, or wired.

**[0062] Header or Packet Header**: The portion of a packet that describes the packet; the meta data concerning the payload, its encoding type and destination.

**[0063] Vox packet**: A proprietary packet that enables the system and method to further refine and optimize the delivery of Messages, Media and other signaling information.

**[0064] Media Payload (or Payload)**: The actual Media portion of a Packet.

### B.3. Media Management

**[0065] Time Shift Delay (TSD)**: The amount of time between the arrival of a Vox Packet and the rendering of the Packet on a Device. The TSD must exceed the **Minimum Time Shift Delay**. The TSD is typically determined by the User's behavior in choosing to review the Messages of a Conversation some time after receipt.

**[0066] Minimum Time Shift Delay (MTSD)**: The time shift delay enforced by the Client to allow for jitter processing using jitter buffer techniques. This causes the system to delay rendering until an adequate number of the packets have arrived to create a usable media stream. The system will typically adaptively adjust the MTSD over time to compensate for variable conditions in the network.

**[0067] Rendering**: Delivering a Media stream to a User in a form suitable for User consumption (e.g., voice, text, graphic display, video, or a combination thereof).

**[0068] Mixing**: The Rendering of one or more Media streams. For example, the Media stream from two Participants of a Conversation may be Mixed when Rendered, creating a User experience similar to a conversation where multiple people are speaking at the same time.

**[0069] Encoding**: The process of translating Media either created by a User (such as voice or video) or otherwise originating on a Device (such as GPS or other sensor data), and converting the media into digital data to be processed by a Client.

**[0070] Adaptive Jitter Buffer**: Jitter buffers or de-jitter buffers are used to counter jitter (i.e., either the arrival of out of sequence packets or the delayed arrival of packets) introduced by packet switched networks, so that the continuous rendering of audio (or video) signals transmitted over a network can be performed without disruption. The data is stored in a buffer before Rendering to allow a reasonably sized buffer of Media to arrive. The Media may be rendered before all the Packets are received, trading off quality for currency. An Adaptive Jitter Buffer is capable of dynamically changing its size to optimize the delay/quality tradeoff.

**[0071] Persistent Infinite Message Buffer (PIMB)**: The PIMB is a storage management system for the storage of time-based Media that performs both the de-jittering of "live" data and the storage and retrieval of archive data. The PIMB further includes the additional attributes of potentially infinite and persistence storage of Media. The PIMB maintains "exact" or full copies of Vox Packets of a Message and Conversations at some or all Participant Devices and/or Servers.

**[0072] Packet Loss Compensation or Concealment) (PLC)**: During Rendering of a Media stream, the PLC component compensates for missing Packets, interpolating the results to present the stream to a reviewer. Missing Packets may be rendered as silence, or information from adjacent Packets may be used to present an interpolated sound or image. The particular method to be used will be dependent on the media, Codecs in use, and other generally known parameters.

### B.4. People

**[0073] User**: A person who is authorized to use the system.

**[0074] Contact**: A record of either a User or non-user of the system. Users typically engage in Conversations with members on their list of Contacts. A non-user is a user that accesses or uses the system using a legacy phone, radio or other non-Client 12 enabled device.

**[0075] Group**: The association of multiple Contacts. Contacts may be selectively added or deleted from a Group. When a Conversation takes place among a Group, all the members of the Group may or may not participate.

**[0076] Channel**: Typically used for tactical communication systems. A Channel is similar to a Group in that it associates multiple Contacts with the Channel.

**[0077] Participant**: A person who is identified as a member of a Conversation. Could be a User or a non-User participant.

### B.5. Conversation Management

**[0078] Time Shifting**: Time shifting is the ability to play any Message at any time after it has been received as determined by the User-recipient. By Time-Shifting, a User may Review a Message: (i) immediately on demand by Rendering immediately after the MTSD; or (ii) time-shifted in a mode of reviewing the Message upon the discretion of the User; (iii) from the archive for searching, reconstructions, etc. of old Conversations; (iv) after a delayed period of time to accommodate the Reviewing of other higher Priority Messages (or Conversations) that need to reviewed first; (v) and/or repeatedly if necessary for the Message to be reheard and understood. In other words, Time Shifting is the ability of a user to render a Message at any time after the system imposed MTSD.

**[0079] Reviewing**: Listening, viewing, reading or otherwise observing the Media content in Messages. Reviewing may take place in either a near synchronous real-time "live mode" or the time-shifted mode.

**[0080] Intention**: Either (i) a User-defined attribute capturing whether the User wants to Review the Messages of a Conversation either as soon as possible or Review the Messages in a time-shifted mode; (ii) implied by a User's behavior; or a combination of (i) and (ii). .

**[0081] Attention**: A user attribute capturing whether the User is Reviewing the Messages of a given Conversation at the moment.

**[0082] Catch Up To Live (CTL)**: A rendering mode that allows a User, who is not at the Head of a Conversation, to Review previous Messages more quickly to "Catch Up To Live" (i.e., the Head of the Conversation). The CTL feature may use any of a number of catch up techniques, such as the faster replay of Messages, the removal of gaps in the Media of the Messages, removal of hesitation particles, etc. When the User has caught up to live, the system seamlessly flows into the live Conversation. This is a very useful feature with conference calls, for example, in situations where a User needs to temporarily shift their attention away from the Conversation, but wishes to hear the entire Conversation upon their return.

**[0083] Catch Up Mode**: A user-configured or pre-configured mode that determines how the CTL process will catch-up (i.e., replay faster, remove silence, and hesitation particles, or a combination thereof).

**[0084] Jump To Live (JTL)**: This feature allows a user to jump from their current position to the Head of a Conversation. A user will typically use the JTL feature when they do not want to Review all of the Messages between their current position in the Conversation and the Head. When the JTL feature is implemented, the user skips over any intervening Messages and starts Rendering the "live" Messages at the head of the Conversation.

**[0085] MCMS Participant Attributes**: A set of attributes, either defined by a User, interpreted by the system from the User's behaviors, assigned by an administrator, or a combination thereof, which define the Intention, Attention, Priority, and rendering preferences of a receiver for a given Conversation. The attributes include, but are not limited to: (i) the Intention of when a receiver would like to render to the Messages of

the Conversation. Possible Intention values include: "now", "time-shifted", "Catch Up To Live" (CTL), "paused", and "never"; (ii) Catch Up Mode, which is a configuration setting which determines how the CTL process should catch the receiver up to live (e.g., play faster, skip silence gaps or hesitations, or play at normal speed); (iii) Time Shift Delay (TSD), which defines how far the receiver's current position in the conversation is from the Head of the Conversation, and (iv) the Priority of the Message with regard to the receiver's other Conversations.

### C. System Architecture

**[0086]** Referring to Figure 1, a block diagram of the telecommunication and media management system according to one embodiment of the invention is shown. The system 10 includes a plurality of Clients $12_1$ through $12_n$, running on Devices $13_1$ through $13_n$ respectively. The Devices 13 communicate with one another over a communication services network 14, including one or more Servers 16. One or more networks $18_1$ through $18_n$, is provided to couple the plurality of Devices $13_1$ through $13_n$ to the communication services network 14. In various embodiments, the networks 18 may be the Public Switched Telephone Network (PSTN), a cellular network based on CDMA or GSM for example, the Internet, a tactical radio network, or any other communication network, or a combination thereof. The communication services network 14 is a network layer on top of or otherwise in communication with the various networks $18_1$ through $18_n$. In various embodiments, the network layer 14 is either heterogeneous or homogeneous. Clients $12_1$ through $12_n$ communicate with one another and with Servers 16 over the networks $18_1$ through $18_n$ and network 14 using individual message units referred to as "Vox packets", which are described in detail below.

### D. Client Architecture

**[0087]** Referring to Figures 2A and 2B, a block diagram of a Client 12 running on a Device 13 is illustrated. As illustrated in Figure 2A, the Client 12 includes Multiple Conversation Management System (MCMS) application 20, a rendering and encoding module 21, and an MCMS applications database 22. As illustrated in Figure 2B, the Client 12 further includes a Store and Stream (SAS) module 24 with a Persistent Infinite Message Buffer (PIMB) reader 26, a PIMB writer 28, PIMB database 30, a data and network quality (DNQS) store 32, and Media driver and encoder hardware 34. The MCMS application 20 and the Store and Stream module 24 communicate with one

another through message handling modules 25a and 25b respectively. The Client 12 further includes an authentication-encryption-security module 40 and a communication protocol module 44.

**[0088]** The module 40 provides authentication, encryption and security services during the transmission and receipt of "Vox" packets to and from the Client 12. The communication protocol module 44 encapsulates Vox packets into the native packets used by the underlying network 18 connected to the Device 13 running the Client 12 when transmitting data and de-encapsulating Vox packets from the native packets when receiving data. With the modules 40 and 44, multi-party end-to-end authentication, encryption and security is provided between Clients 12. Messages are authenticated, encrypted and secured across the networks $18_1$ through $18_n$ and network 14, from a first sending Device 13 to second receiving Device 13.

### D.1.1. The MCMS Database

**[0089]** The database 22 stores and manages the persistent meta data for a number of entities in the system 10, including Contacts and Participants, Conversations and Messages (live and stored), and default Priorities, and information regarding the Servers 16. In addition, the MCMS database 22 stores the moment-to-moment operational data of a User's Conversations, presence, and status, as well as that of all the Participants conversing with the User or on the User's Contact list. For example, with regard to Conversations and Messages, the database 22 keeps track of status information, such as what Messages of a Conversation a User has or has not Reviewed, Priorities, and Catch Up To Live status for each Conversation in which the Client 12 is a Participant, the presence and status of all Participants, and other network and other system management data.

### D.1.2. The MCMS Application

**[0090]** MCMS application 20 supports the different Voxing modes of engaging in conversations and/or managing multiple conversations using a variety of media and data types (voice, video, text, location, data, etc.). Users engage in Conversations by sending Messages to designated recipients using their Client 12 enabled Devices 13. Depending on preferences and Priorities, the recipient might Review the Message in real time, or they might simply be notified that the Message is ready for Reviewing. Users can transition from a series of back and forth Messages, which are Reviewed in a time-

shifted (or voice messaging) mode or in a near synchronous, full duplex conversation (similar to standard "live" phone calls) and then back to voice messaging again. The MCMS application 20 allows a User to control their interactions with their most important Conversations in real-time without missing any Messages in other ongoing Conversations. For example, the MCMS application 20 notifies a User of urgent or high priority communications from a Conversation that they are not currently Reviewing. MCMS application 20 also enables all Messages from all Conversations to be stored for later retrieval so they can be reviewed at any time.

**[0091]** In accordance with various embodiments, there are several different operational modes of the MCMS application 20, including MCMS-Consecutive (MCMS-C) and MCMS-Simultaneous (MCMS-S), which support the consecutive and simultaneous rendering of Messages respectively. Each of these embodiments is described in more detail below. Unless specifically specified, the term "MCMS" is intended to generally mean the MCMS application 20, which includes the aforementioned different modes.

**[0092]** The MCMS application 20 is a multi-tiered architecture that includes a number of modules and services. The modules and services include the MCMS Database Module 20a, the SAS Services Module 20b, the Messaging and Signaling Services Module 20c, the User Interface Application Programming Interface (API) 20d, the User Interface Module 20e, the Conversations/Messages Management Services 20f, the Priorities Services 20g, the Contacts Service 20h, the Presence/Status Services 20i, and the Messages/Signals Services 20j.

#### D.1.2.1 The MCMS Database Module

**[0093]** The MCMS database module 20a is a service module that manages all function calls necessary for the MCMS application 20 to access the MCMS database 22.

#### D.1.2.2 The SAS Services Module

**[0094]** The SAS Services module 20b includes a set of function calls that enable communication and coordination between the MCMS application 20 and the Store and Stream module 24, and which are passed back and forth through the message handling modules 25a and 25b respectively. The set of function calls enable both the MCMS application 20 and the Store and Stream module 24 to operate as necessary to implement the various Voxing functions when invoked by Users and/or as dictated by network conditions. Some of the functionality performed by the SAS Services module

20b includes maintaining and communicating the status of Message transmissions and Message acknowledgments, the instructions for rendering of Messages, and the status and presence of Users.

#### D.1.2.3 The Messaging and Signaling Services Module

**[0095]** The Messaging and Signaling Services module 20c runs on both Clients 12 and Servers 16 and enables communication between the Client 12 and the Servers 16 of the system 10. This communication, which includes messages, data and other signals, allows the Client 12 and the system 10 to track and administer communications, network status, Users, and User status. Types of messages and signals sent between the Message and Signaling Services modules 20c running on the Clients 12 and the Servers 16 include, for example, network availability of Users, tracking of Messages that the Server 16 has sent to the Client 12 (possibly including a "high water mark") to determine if an entire message or some portion of a message is missing, (e.g., a sequence number per Participant per Conversation created by the "generating" Client), whether a user is speaking or Reviewing Messages of a given Conversation, where a User is with respect to the Head of a Conversation, or when a Participant is no longer Reviewing a Conversation live. These are examples a few of the many types of messages and signals sent between the Message and Signaling Services modules on the Clients 12 and Servers 16 and in no way should be construed as limiting the invention.

#### D.1.2.4 The User Interface API

**[0096]** The User Interface API 20d is a module that defines a set of function calls that define the programming interface between the User Interface module 20e and the underlying services of the MCMS application 20. The User Interface API 20d supports general-purpose methods such as UI application support, and all function calls necessary for a User Interface to operate the MCMS Application 20. In various embodiments, the User Interface API 20d enables the Client 12 to support a wide variety of user interfaces and device types, such as Adobe Flash-based and/or Microsoft Windows applications, cellular or mobile phone devices, PSTN devices driven with tones, a voice user interface (VUI), and physical radio communication interfaces. In various embodiments, the User Interface API module 20d enables the design of both highly flexible and highly constrained user interfaces to support the functionality of the MCMS application 20.

#### D.1.2.5 The MCMS User Interface Module

**[0097]** The MCMS User Interface module 20e supports the operation and functions of the audio and video user interface of the Client 12. The User Interface module 20e

supports a host of user interactions and can be implemented with various interaction mediums, such as, an array of graphical user interface screens, an Audio/DTMF interface, or voice user interface on the Device 13, all of which enable a User to interface with the system 10. A partial list of User interactions that are supported include, for example, functions to: log-in; manage, join, and monitor Conversations; control Conversation rendering; manage Priorities; and requests to review archived Conversations. It should be noted that this list is exemplary and in no way should be construed as limiting the invention.

#### D.1.2.6 Conversation/Message Management Services

**[0098]** The Conversation/Message management services 20f is a module which defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for the User to manage the receipt and Review of transmitted and received Media (e.g., voice or video content Messages) between the participants of a Conversation. The Messages are organized into Conversations. Media that is sent or received by the Device 13 running the application 12 is available for immediate Review while being received. The received Media is also recorded for Review in a time-shifted mode, Conversation management, and archival purposes. In an alternative embodiment, Messages or Conversations can be optionally marked for transience, specifying their desired retention requirements (e.g., some Messages will not be retained or stored beyond the requirements for immediate rendering). In yet another embodiment, Media can be optionally marked for review in a time-shifted mode only and cannot be reviewed immediately upon receipt.

**[0099]** The Conversation/Message management services 20f further enables, for each current or ongoing Conversation of the User, the sending of Media to a receiving Client 12 at any time, and the receiving Client 12 seamlessly associates these Messages with the appropriate Conversation, regardless of the actions or inaction of the receiver.

**[00100]** With the Conversation/Message management services 20f, all Conversations are essentially asynchronous. If two Users are actively engaged in a given Conversation and the User controlled delay between transmissions is minimal, the experience will be one of a synchronous full duplex conversation, as with current telephone or VoIP conversations. If either User delays their participation, for whatever reason, the Conversation drifts towards an asynchronous voice (or other Media)

messaging experience. In alternative embodiments, Conversations can be optionally Tagged as asynchronous Messages only or synchronous Messages only. In either of these cases, the Conversation cannot drift between the two modes, unless the Tag is reset. After the Tag is reset, the Conversation again may flow between near synchronous (i.e. live or real-time) and asynchronous (i.e., time-shifted or voice messaging) modes.

**[00101]** The Conversation/Message management service 20f processes the transmission and receipt of Messages in a progressive fashion. When transmitting, Media may be created while Messages are simultaneously encoded, stored and transmitted. In other words, the transmission of Messages may occur simultaneously with the generation of Media by the User (i.e., while speaking into their Device 13 or generating video). On the receiving side, the receipt, storage, and Rendering of Messages also all occur progressively. Messages do not need to be completely received before they can be Rendered. The Rendering of Messages may occur at the same time Messages are being delivered, right up to the MTSD. Further, the service 20f is also capable of the simultaneous transmission of outgoing Messages and Rendering of incoming Messages. The progressive nature of the service 20f allows a User to be engaged in a live Conversation while storing and streaming the media of the Conversation for later retrieval and review as well other functions described herein.

**[00102]** The time-shifting of Messages by the Conversation/Message management services 20f allows a User to "Catch Up To Live" on a Conversation if they missed earlier Messages or were involved in another Conversation. This time-shifting process eliminates the need for Users to broadcast a request to their entire Group or Channel to have Messages repeated. Older Messages may be replayed at any time at potentially higher speeds to save time. Users may easily skip forward and backward through their Messages and within individual Messages. The Reviewing process may be configured on a Message-Priority basis to potentially skip lower priority Messages.

**[00103]** In one embodiment, the Conversation/Message management service 20f also identifies Messages by a specific Participant (speaker) and, by default, mixes Messages of a Conversation that were delivered at the same time (MCMS-S). In an optional embodiment, a User could Review the transmissions of different Participant speakers of a Conversation separately.

**[0100]** The Conversation/Message management module 20f further allows Conversation sharing among Participants, who can be added to an active or archived Conversation. In one embodiment, an added Participant to a Conversation is provided access to the Conversation and has the ability to retrieve the previous Messages of the Conversation for Review. In an alternative embodiment, the added Participant is provided access to the Messages of the Conversation only from the point where the new Participant joined, and not any previous Messages of the Conversation.

**[0101]** The Conversation/Message management module 20f is also responsible for managing the functions used to control all rendering tasks performed by the Store and Stream module 24. These tasks include rendering Media (i.e., voice, video, etc.) appropriately for the Device 13 running application 12. These rendering tasks include, but are not limited to, the rendering of Mixed Messages (i.e., overlapping messages), as well as rendering according to user-defined criteria, such as playing faster, catching up to live, removing silence, removing hesitation particles, frequency shifting, and the ability to apply independent gain control to individual senders in a multi-party conversation.

#### D.1.2.7 Priority Services

**[0102]** The Priority service 20g is a module that defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for a User to manage the Priority of the consecutive Conversations (i.e., MCMS-C) in which the User is engaged. When a User participates in a number of consecutive live Conversations, the User is required to prioritize the Conversations. Issues arise when Messages of different Conversations are ready to be rendered at the same time. An algorithm is used to determine the order in which the Messages are rendered that considers the availability of Messages to be rendered and the Priorities set by the User. The algorithm determines that the available Messages with the highest priority are rendered first while any concurrently available Messages are time shifted automatically just enough to allow for the rendering of the higher priority Message. As rendering time becomes available, the system will automatically render the time-shifted messages according to the User's Priorities.

#### D.1.2.8 Contacts Services

**[0103]** The Contacts services 20h is a module that defines a set of functions that manage the data structures and processes responsible for managing and retaining all information needed for authenticating and associating one or more Contacts with Conversations. When sending a Message as part of a Conversation that is associated with a number of Contacts, all of the Contacts receive the Message.

#### D.1.2.9 Presence/Status Services

**[0104]** The Presence/Status services 20i is a module that defines a set of functions that maintain the data structures and processes responsible for managing and sharing presence and status information between certain Users and/or non-users of the system. In various embodiments, the presence and status information is maintained for all User and non-users engaged in the Conversations the User of the Client 12 is engaged in, all Users and non-users in the Contacts list, or Users within a predefined domain (e.g., the members of a corporation or other organization). These examples are merely illustrative and should not be construed as limiting. The Presence/Status services 20i module may manage and share presence and status information on any defined set of Users and/or non-users.

**[0105]** The Presence/Status service 20i enables Users to monitor the status of other User's Intentions, Attention, and their Time Shift delay on any given Conversation (i.e., how far back they are from Reviewing the Messages of the Conversation live). In one embodiment, privacy controls are provided concerning availability of Presence and Status data. The Presence/Status module 20i further controls the data that enables the system 10 to deliver Messages that match the behavior and Intentions of Users. For example, a User may indicate their Status by designating an Intention to either Review or not Review a Conversation live. In response, the Presence/Status services 20i issues commands that cause the rendering of Messages either "live" or time-shifted, in accordance with the Intention of the User. In addition, the Intentions of Users are shared with the other Participants of the Conversation. The service 20i is also capable of inferring other Status values from the User's behaviors. The Presence and Status information is also used to optimize network traffic and bandwidth, as described in more detail below.

#### D.1.2.10 Messages/Signals Services

**[0106]** The Messages/Signals services 20j is a module that defines a set of functions that manage the data structures and processes responsible for messaging and signaling Users of the system 10, using special messages or audible tones. The special messages or tones may include for example an indication if a Message or Messages are live or time-shifted, whom the Message(s) are from, Priority, and other factors. The Message/Signal service 20j further has the ability to (i) signal the presence or absence of Users on the network, as well as the ability to notify if one or more Users are no longer actively Reviewing the Messages of a Conversation; (ii) "ring" or otherwise notify another User to get their attention when they are paying attention to another Conversation or not paying attention to their Device 13 at all; (iii) leave a Message for Users currently not on the network 18 to immediately review the Message the next time the individual re-connects to the network 18; (iv) generate an audible or visible feedback that alerts the sender that a sent Message was not received by recipient(s), generate a confirmation when the Message has been received by the recipient(s), and/or a confirmation indicating when the Message has been Listened to by the recipient(s); and (v) implement a Priority scheme where individuals on a Conference or tactical call may be notified that their attention is immediately needed on the call. This indication may convey multiple levels of urgency and an acknowledgement of some kind by the recipient.

#### D.1.2.11 Rendering and Encoding

**[0107]** The rendering and encoding module 21 is responsible for performing all rendering tasks for the MCMS application 20. These tasks include rendering Media appropriately for the device 13 running application 12.

### D.2 The Store and Stream Module

**[0108]** The Store and Stream module 24 supports a number of functions and performance attributes, which are described below.

**[0109]** With the Store and Stream module 24, Message transmission is essentially "full-duplex", enabling any party to send a Message at any time, even while another party is also sending a Message, or if the other party is unavailable or otherwise engaged. The Store and Stream module is able to render messages as in a live PSTN or VoIP call or

deliver them for time shifted messaging modes. It is able to optimize transmission and control Rendering according to the desires of the User.

**[0110]** The Store and Stream module 24 maintains connectivity with all target recipients (e.g., Servers 16 or other Devices 13) on the underlying network 18, manages all message, signal, and media transmissions, and optimizes the delivery speed and bandwidth usage across the network 18 to meet a User's immediate performance requirements, while managing network quality and capacity. The module 24 adapts and optimizes Media delivery commensurate with the quality and capacity of the underlying network 18. When insufficient underlying network resources are available, the quality of the transmitted Media streams can be degraded. As bandwidth becomes available, the quality of the transmitted Media streams may be increased. In addition to tradeoffs of Media quality, the Store and Stream functionality can make tradeoffs in the amount of Media transmitted in each packet based on Users' intentions to render data in real time as described below.

**[0111]** By dynamically controlling the delivery rate of Media based on the conditions of the underlying network 18, the Store and Stream module 24 is optimized to deliver time-sensitive Media that is "good enough" to Render upon receipt, and the guarantee eventual delivery of exact or full copies of the Media for archival purposes through a background process of requesting retransmission of missing, low quality, or damaged packets. As long as sufficient network resources exist to meet minimum Media quality levels, this retransmission does not impede the Rendering of live call Media. The Clients 12 of the system 10 are thus designed to bridge the performance gap between the delivery of an exact or complete copy of the Media at the expense of substantial potential latency versus the quick delivery of Media, but with no guarantees of completeness. In the context of this application, the term "good enough" means that the quality of the Media is sufficient so that when it is rendered, it is intelligible. The notion of "good enough" is therefore subjective and should not be construed in absolute terms. For example, the quality level of certain Media to be good enough may vary depending on the type of Media, circumstances, and other factors.

**[0112]** The Store and Stream module 24 further persistently stores all Media created by or otherwise originating using a Device 13 or received over the network 18 from other Device 13 and/or users. There are several significant advantages of storing this Media on the Device 13 running the Client 12: (i) it enables Users to leave a Message for

another party, even when the sender and/or the recipient has either unavailable or poor network connectivity. In the case of insufficient bandwidth, the Message will be transmitted as fast as available bandwidth can be effectively used. In the case of no connectivity, the Message is queued for transmission as soon as network connectivity becomes available, resulting in a time-shifted delivery; (ii) the User has the ability to pause, replay, fast-forward, and Catch-Up-To-Live with an ongoing Conversation, as well as retrieve and review the archived Messages of previous Conversations; and (iii) it enables the optimization of data payloads over the system 10 and improves system resilience against network bandwidth and connectivity problems that may occur from time to time.

**[0113]** The Store and Stream module 24 is also responsible for: Mixing Messages as appropriate to create overlapping Messages (generated by the normal overlap of speakers in a Conversation or background noise), simulating an actual Conversation where multiple parties are speaking; rendering transcriptions or translations of audio media; adjusting the rendering of Media according to a number of User-defined criteria including playing faster, removing silence gaps between spoken words, removing hesitation particles, and frequency shifting; and the ability to apply independent gain control to individual senders in a multi-party Conversation; as well as other potential Rendering options.

**[0114]** The Store and Stream module 24 manages control and informational messaging between itself and MCMS.

### D.2.1 The Persistent Infinite Message Buffer (PIMB)

**[0115]** The Persistent Infinite Message Buffer or PIMB 30 is a set of indexed (i.e., time-stamped and sequentially numbered) Media payload data structures and a system for their storage and retrieval. In one embodiment, the data in the PIMB 30 is arbitrarily persistent, meaning it is available virtually forever or at least until the system runs out of storage. Various retention rates and strategies may be employed to make effective use of storage resources. Many possible implementations exist for the physical storage implementation of the PIMB 30, including, but not limited to: RAM, Flash memory, hard drives, optical media, or some combination thereof. The PIMB 30 is also "infinite" in size, meaning the amount of data that can be stored in the PIMB 30 is not inherently limited. This lack of limit is in comparison to existing jitter buffer technology that

discards data as soon as it is rendered. In one specific embodiment, the PIMB 30 may be implemented using a small and relatively fast RAM cache memory coupled with a hard drive for persistent storage. As the physical storage capacity of the PIMB 30 is exceeded, the data is maintained on the Server 16 (as described below) for later retrieval on demand. User criteria or a replacement algorithm, such as least-recently-used, or first-in-last-out, is used to control the actual data stored in the PIMB 30 and the data that is maintained on the Server 16 or archived at any point in time. The PIMB 30 further provides the attributes of file system storage and the random access attributes of a database. Any number of Conversations, regardless of their duration or the number of Messages in each, may be stored and later retrieved for Review. In addition, the meta data associated with the Messages of a Conversation, such as its originator and its length, may be also stored in the PIMB 30. In alternative embodiments, the indexed Media payloads and other data can be stored for a designated period of time (e.g. 30 days). Once the age of the media exceeds the designated period, the payloads and data are discarded. In another embodiment, payloads can be discarded based on the sender and/or the recipient of the Message containing the payload, or the topic of the Conversation or Messages associated with the payload. In yet other embodiments, payloads and data may be marked for transience, meaning the Messages will not be stored in the PIMB 30 beyond the requirements for immediate Rendering.

#### D.2.2 The Data and Network Quality Store

**[0116]** The data and network quality store (DNQS) 32 is a data store for storing information regarding Media payloads and Vox packets that are either read from or written to the PIMB 30.

#### D.2.3 The PIMB Writer

**[0117]** The PIMB writer 28 writes data to the PIMB 30 for two basic purposes. The PIMB writer 28 writes data from a Media capturing device (e.g., a microphone or camera) on the Device 13 running the Client 12 ("Encode Receive"). The PIMB writer 28 also writes data received over the network 18 from other Clients 12 into the PIMB 30 ("Net Receive").

##### D.2.3.1 Encode Receive

**[0118]** For capturing Media from the Device 13, the PIMB writer 28 includes Encoder Receiver 28a and a Data Storer 28c. When a User speaks into the microphone or

generates video images with the Device 13 for example, the hardware 34 receives the raw audio and/or video signals and provides them to the Encoder Receiver 28a, which encodes the signals into indexed media payloads (hereafter sometimes simply referred to as "payload" or "payloads"). The Data Store 28c stores the payloads into the PIMB 30. Other types of Media, such as sensor data is converted into payloads in a similar manner.

#### D.2.3.2 Net Receive

**[0119]** For storing Media received over the network 18 into the PIMB 30, the Net Receive function of PIMB writer 28 includes a Network Receiver 28d, a Data Bufferer 28e, a Data Storer 28f, a Data Quality Manager 28g, and a Network Quality Manager 28h. The Network Receiver 28d receives Vox packets over the network 18. The Data Bufferer 28e places the received Vox packets into their proper sequence and prevents the Rendering of the incoming Vox packets for at least the Minimum Time Shift Delay (MTSD) delay. The Data Storer 28f transforms the packets into indexed media payloads and stores the indexed media payloads in the PIMB 30. As the payloads are stored, the Data Quality Manager (DQM) 28g notes any missing or defective packets. If packets are missing or defective, the DQM 28g schedules a request for retransmission over the network 18. The sending device replies by resending the missing or defective packets. Eventually these packets are converted into indexed media payloads and stored in the PIMB 30. By retrieving the missing or defective packets, an "exact" copy of a sender's Message is eventually stored in the PIMB 30. The retransmission of missing and/or defective packets does not delay the rendering of Messages in real time, provided that the packets that have been delivered are of "good enough" quality and quantity. Retransmission requests may be delayed by the DQM 28g if insufficient network resources are available to support both the new "live" data as well as the retransmission.

### D.2.4 The PIMB Reader

**[0120]** The PIMB reader 26 reads data from the PIMB 30 for two basic purposes. The PIMB reader 26 accesses the PIMB 30 when data is to be rendered ("Render") for the local Client 12. Data is also read from the PIMB 30 when data is to be transmitted ("Transmit") by the Client 12 over the network 18.

#### D.2.4.1 Render

**[0121]** For the rendering of messages on the Client 12, the PIMB reader 26 includes a Data Prioritizer 26a, a Data Retriever 26b, a Packet Loss Compensation/Interpolator ("PLC/Interpolator") 26c, a Data Mixer 26d and a Data Renderer 26e. The Prioritizer 26a prioritizes the data to be rendered by building an ordered queue of messages that could potentially be Rendered. It uses User configured Priority for the rendering of consecutive Conversations (MCMS-C). In addition, the Data Prioritizer uses the availability of media data to render within the limits imposed by the MTSD, the User's current Attention, and the User's defined and implied Intentions. The Data Retriever 26b retrieves the prioritized indexed media payloads from the PIMB 30. The PLC/Interpolator 26c performs packet loss compensation and interpolation on the retrieved payloads, using known packet loss compensation and interpolation algorithms. The particular method to be used is dependent on the media Codecs in use, and other well-known parameters. The Mixer 26d is used to appropriately mix concurrent data streams from multiple Messages of a single Conversation together. For example, if two or more Participants of a Conversation are speaking at the same time, the Mixer 26d mixes the Messages, creating the effect of both Participants speaking at the same time. In an alternative embodiment, the User has the option of Reviewing the multiple streams from one Participant at a time. If only one Participant in the Conversation is speaking, the Mixer 26d may simply pass the single Message stream, without performing any mixing. The Renderer 26e takes the data from the Mixer module 26d and converts it into a form suitable for the hardware driver 34. The hardware 34 then drives either the speaker or video display of the Device 13, depending on the type of Media, creating voice, video or some other audible and/or visible notifier on the Device 13.

#### D.2.4.2 Transmit

**[0122]** To prepare messages for transmission from the Client 12 over a network 18, the PIMB reader 26 includes a Data Prioritizer 26f, a Packet Quality Manager (PQM) 26g, a Data Retriever 26h, Packetizer 26i, a Transmitter 26j and an Acknowledger 26k. The Data Prioritizer 26f prioritizes the Messages for transmission over the network 18. Priority is determined using the MCMS Participant Attributes related to payloads available for transmission, network connectivity and bandwidth conditions, and the Intentions of Users beyond the next hop to either Render live or time-shifted, and, in

some embodiments, possible optimizations of transmission bundling where multiple packets to any given next network hop are available. The prioritized packets are then optimized, using the PQM 26g, which assures the timely delivery of "good enough" data quality for live Messages, while minimizing real-time bandwidth, as described in more detail below. The Data Retriever 26h retrieves the appropriate payloads from the PIMB 30. The Packetizer 26i assembles the payloads into Vox Packets, which are then transmitted by the Transmitter module 26j over the network 18. When the recipient receives the Vox packets, an acknowledgement is sent back to Acknowledger 26k over the network 18 for notifying the sending User that a Message has arrived at its destination.

### D.2.5 The Packet Quality Manager

**[0123]** The PQM 26g has several optimization goals: (i) the timely delivery of an adequate copy of time-sensitive Media ("as soon as possible to be good enough" for Rendering); (ii) the efficient use of available bandwidth, meaning using the optimal transmission frequency, payload quality, and packet size for the underlying network; and (iii) the ability to dynamically adjust or make changes in the transmission frequency, payload content, payload quality, packet size, etc. as network conditions change.

### D.2.6 The Network Quality Manager

**[0124]** On the receiving side of a network transmission is the Network Quality Manager 28h (NQM). The NQM is responsible for observing specific properties of network performance for each sender that has sent media to the Client 12, comparing expectations of jitter, loss, and throughput to their actual values. This is used to compute a Network Quality Rating (NQR) for every sender. This NQR is used to indicate sender availability and Conversation live-ness to the User of the receiving device.

### D.2.7 The Data Quality Manager

**[0125]** The Data Quality Manager 28g measures the quality of the data being received over the network by observing packet loss, jitter, and throughput. The DQM 28g uses these measurements for three purposes: (i) to send receipt reports back to the sender; (ii) optionally using those receipt reports to request retransmission of certain data; and (iii) making these measurements available to the NQM 28h.

## E. Server Architecture

**[0126]** Referring to Figure 3, a block diagram of the application 78 that runs on the Server(s) 16. The application 78 is similar to the Client application 12 in many regards and includes an MCMS server application 80, an MCMS database 82, a store and stream module 84, a PIMB 85, a data and network quality store (DNQS) 86, MCMS-SAS message handling modules 87a and 87b which manage messages and signals back and forth between the MCMS server application 80 and Store and Stream module 84, an archive/retriever 88, and an archive 89. The application 78 further includes an authentication-encryption-security module 40 and a communication protocol module 44.

**[0127]** The MCMS server application 80 is a multi-tiered architecture including a MCMS Database Module 20a, a Store and Stream (SAS) Module 20b, a Messaging/Signaling Module 20c, Conversations/Messages management services 20f, Priority services 20g, Contacts (including User and Authentication) services 20h, Presence/Status service 20i, and Messages/Signals services 20. The aforementioned modules and services of the application 78 are similar or the same as the modules and services having like reference numerals as the Client 12, and therefore are not described in detail herein, except for one notable exception. In various embodiments, the MCMS server application 80 and Store and Stream module 84, including the MCMS database 82, is configured to support many Users in one instance of the application. The one instance may be further configured to support multiple domains, where each domain is defined as a group of Users (i.e., a corporation or other group of Users belonging to a common organization). This architecture allows each application 78 on Server 16 to serve multiple users (or domains), where each user (or domain) is not visible to other Users. This partitioning is referred to as "multi-tenancy".

**[0128]** The server Store and Steam 84 module performs the functions of Net Receive and Transmit. For the Net Receive function, the module 84 includes a Net Receiver 28d, Data Bufferer 28e, a Data Storer 28f, a Data Quality Manager (DQM) 28g, and a Network Quality Manager 28h. For Transmit functions, the module 84 includes a Data Prioritizer 26f, Packet Optimizer 26g, Data Retriever 26h, Packetizer 26i, Transmitter 26j and an Acknowledger 26k. The aforementioned elements of the Store and Stream module 84 are similar or the same elements having like reference numerals as the Client 12, and therefore are not described in detail herein.

**[0129]** Since the Server 16 does not directly interact with Users, the encoding and rendering functions provided in the Store and Stream module 24 of the Clients 12 need not be present. The MCMS application 80, when running on Servers 16, does not interact directly with Users. Consequently, the user interface and user interface API modules and services 20e and 20d are not needed.

**[0130]** The application 78 on each Server 16 potentially serves multiple tenants, meaning it serves multiple Users of the system 10. The PIMB 85 of the server application 78 is therefore significantly larger and is used to store the media payloads of multiple Users, as opposed to the PIMB 30, which is used to store just the generated or received payloads of only a single User. The main purpose of the Store and Stream module 84 is to receive Messages transmitted by the Clients 12 and transmit Messages to other Clients 12. As Messages are received, they are stored in the PIMB 85 and transmitted to the next Server 16 (i.e., the net "hop") of the network layer 14 along the path to the intended recipient(s), or to the recipient(s) directly depending on the system configuration. The archive-retriever 88 is responsible for archiving the media payloads stored in the PIMB 85 in archive 89. As the physical space in the PIMB 85 is exhausted, media payloads in the PIMB 85 are moved to the archive 89, which is a mass storage device. In various embodiments, the payloads stored in the PIMB 85 may be archived in accordance with User defined criteria and/or any known replacement algorithm, such as first-in-first-out (FIFO) or least recently used (LRU). It should be noted that only a single Server 16 is illustrated in Figure 1 for simplicity. It should be understood that in actual embodiments, multiple servers or a "server farm" may be used for a network with a large number of Users.

**[0131]** The terms "persistent" and "infinite" used to describe the PIMB 30 and the PIMB 85 should not be construed literally as absolute terms. A User may wish to indefinitely store some Messages that are considered important. In other situations, such as a casual chat between two friends, the Messages may be deleted after a certain period of time to save space. According to various embodiments of the invention, different retention policies may be used, either set by the system 10 or configured by the User. The use of the word "infinite" refers to the lack of any preset time boundaries enforced by the PIMB. This is contrasted with current jitter buffer systems, which discard media after it has been rendered. The terms persistent and infinite should therefore be broadly

construed to mean that the PIMB 30 and PIMB 85 have no internal limitations on the time ranges and quantities of Messages that can be stored therein.

**[0132]** There are a number of advantages to archiving the Messages of Conversations in a persistent storage medium. Voice messages and other Media can be organized, indexed, searched, transcribed, translated, and Reviewed as needed. Voice, as well as other Media, therefore becomes an asset that can be managed both by Users and organizations. These Media assets have value to corporations, first responders, police and fire departments, as well as the military.

### F. The Vox Protocol and Indexed Media Payloads

**[0133]** As noted above, the Vox protocol is used by the Store and Stream module 24 to support all facets of payload transmission, storage, and optimization. The Vox packet is a structured message format designed for encapsulation inside a transport packet or transport packets of the underlying technology of the network 18. This arrangement significantly improves the flexibility of the system 10. By embedding the Vox packets into existing transport packets, as opposed to defining a new transport layer for "Voxing" applications, the system 10 takes advantage of current packet based communication networks running over the existing telecommunications infrastructure. A new network infrastructure for handling the Vox packets therefore need not be created to take advantage of all the benefits of the system and method described herein.

**[0134]** Referring to Figure 4A, the general format structure of a Vox packet 95 is illustrated. The format of the Vox packet 95 includes fields for type, sub-type, length, and payload. The different types of Vox packets include authentication, signaling, media payload, media multiplex (one message), and media multiplex (multiple messages). The sub-type field is used to designate different types of authentication, signaling or media type messages. Possible sub-types for authentication Messages include those necessary for key exchanges and Authentication. Possible sub-types for signaling Messages include registration, routing, message set-up, and network management. Possible sub-types for Media messages include different Codec styles and different payload aggregation techniques. The length field defines the overall length or size of the payload. The payload field contains the actual payload or media of the packet 95.

**[0135]** Referring to Figure 4B, a diagram illustrating a Vox packet 95 encapsulated in an exemplary protocol used by the network 18 is shown. In this example, the Vox packet 95 is embedded in underlying UDP, IP and Ethernet transport packets 96 respectively. In this manner, the Vox packet 95 can be transported across underlying UDP, IP and Ethernet layers of the network 18. This is a standard protocol encapsulation technique used by packet networks.

**[0136]** Referring to Figure 4C, a diagram illustrating a media multiplex Vox packet 95 encapsulated in UDP, IP, and Ethernet 97 is illustrated. In this example, the Vox packet 95 includes a Media type field, a Media sub-type field, a length field, a message ID field, a time stamp field, a sequence ID field, and a Media payload field.

**[0137]** Referring to Figure 4D, the format of an indexed media payload 98 is illustrated. The indexed media payload includes a sub-type field, a length field, a message identifier (ID) field, a time-stamp field, a sequence identifier (ID) field, and field for the Media payload.

**[0138]** The encapsulation of Vox packets 95 into the transport packets of the underlying network allows the Media, Messages and Conversations to each be defined by a number of attributes.

**[0139]** When Media is created or otherwise originated on a Device 13, it is typically time-based, meaning it changes in some meaningful way over time. As a person engages in a Conversation for example, their spoken words are strung together into sentences or statements, which vary over time, and the resulting data (streams and packets) will maintain the same variance over time. Similarly video (as opposed to a still photo) as well as GPS or other sensor data will vary over time. Regardless of the type or how it is originated, the Media is segmented and placed into the payloads of a plurality of Vox packets 95. The packets are then continually, stored, transmitted, received, stored and rendered in streams (i.e., streaming media) at the transmitting and receiving Devices 13 respectively. Since each packet 95 is indexed, time-stamped, and given a sequence identifier, the individual packets can be segmented into Messages. By sequentially threading the individual Messages together, Conversations may be constructed.

**[0140]** One further unique aspect of the system 10 is that the media payloads generated by a Client 12 are stored in multiple locations. Not only are the payloads stored in the PIMB 30 of the generating Device 13, but also in the PIMB 85 of the Server(s) 16 and

the PIMB 30 of the receiving Devices 13. This basic feature enables or makes possible much of the Voxing functionality described above and provides the system 10 with both resilience and operability, even when network conditions are poor or when a Participant of a Conversation is not connected to the network.

### G. Interoperability With Underlying Telecommunication Protocols

**[0141]** The system 10 is intended to run or be layered over a variety of existing communication networks 18, such as the Internet, fixed PSTN type circuit networks, and mobile or cellular phone networks, or a combination thereof. The system 10 is designed around the concept of moving many small units of information (i.e., the Vox packets) between different Clients 12 and Servers 16 in the system 10. While Vox packets may vary in size, depending on their function and payload, they all appear to be the same kind of data to the underlying network layer. In one embodiment, the system 10 has been designed and optimized for IPv4 networks such as the Internet, but other types of networks may be supported as well. For the purposes of this document, the term "IP" should be taken to mean IPv4, IPv6 or any other current or future implementation of the Internet Protocol.

**[0142]** Referring to Figure 5, a diagram of a Client 12 running on Device 13 and communicating with a Server 16 over a shared IP network 100 is shown. As illustrated, the Client 12 is coupled to the shared IP network 100 through a first Internet service provider A and the Server 16 is coupled to the shared IP network 100 by a second Internet service provider B. During communication, the Vox packets 95 (designed "VP" in the figure) are encapsulated within UDP/IP packets and then interleaved among other IP protocol packets as is well known in the art and transmitted across the shared IP network 100 from the Client 12 to Server 16, or vice versa. As is well known, each lower packet layer encapsulates the entire packet of the layer immediately above it. Packets can also be sent in a similar manner between two Servers 16. Messages are sent from one Client 12 enabled Device 13 to another over a shared IP network 100. At each hop, the Vox packets 95 are embedded in the underlying IP protocol and transmitted, until they reach the target destination.

**[0143]** The diagram of Figure 5 is merely exemplary, showing only a single Client 12 and Server 16 connected to the network 100 for the sake of illustration. In actual embodiments of the system 10, a large number of Clients 12 and one or more Servers 16

are typically connected to the shared IP network 100. It is also useful to note that the Client 12 and Server 16 do not have exclusive use of the IP network 100. In the example shown, an HTTP client, which is coupled to the network 100 through Internet provider A, can send packets back and forth with an HTTP server, coupled to the network 100 through a third Internet provider C. The system 10 does not control the manner in which the VPs embedded in the IP packets traverse the network 100. Rather, all packets that traverse and share the network 100 do so in accordance with the standard procedures of the underlying shared IP network 100.

**[0144]** Referring to Figure 6, a "circuit" based network 104 such as a GSM mobile phone network is illustrated. The circuit network 104 is coupled between Client 12 running on Device 13 and Server 16. Once a circuit is established between the Client 12 and Server 16, the system 10 layers Vox packets (VP1, VP2, VP3, VP4, VP5, etc.) onto the underlying packets used by the network 104 and transmits them across the network 104, creating "virtual Vox" circuit. The Vox Packets sequentially traverse the circuit network 104, typically with spacing or framing data as is well known in the art for transmitting data over a circuit network. In addition, packet construction parameters, such as the payload size and the number of header fields included may be used to exploit the lack of per-packet overhead and to increase speed and/or efficiency of data transfer across the network 104. It should be noted again that for the sake of simplicity, only a single Client 12 and Server 16 are shown connected to the network 104. It should be understood, however, that additional circuits between Clients 12 and Servers 16 as well as other components may be established concurrently through the network 104. The network 104 is therefore not dedicated for the transmission of Vox Packets, but rather may be shared with other types of network traffic.

**[0145]** Referring to Figure 7, a diagram illustrating communication between a first Client 12A enabled Device 13A associated with a first network A and a second Client 12B enabled Device 13B associated with a second network B is illustrated. The networks A and B further each include gateway Servers 16A and 16B respectively. The Gateway Server pair 16A and 16B facilitate communication between the two networks A and B, allowing the Devices 13A and 13B to communicate with each other. In various embodiments, the networks A and B could each be any type of network. For example, each network A and/or B could be an IP network, a circuit type network, or a wireless or cellular network (i.e., CDMA, GSM, TDMA, etc.). The Servers 16 that straddle the two

networks A and B are considered gateway servers because they route traffic or serve as a "gate" between the two networks.

**[0146]** With the system 10, there are a several basic network interaction considerations to optimize system performance. These considerations include factors such as resolving the underlying address to which the Vox packets 95 are to be sent, the integrity of any sent Vox packets, and the management of the Maximum Transmission Unit (MTU) of a single Message that may be sent across a given network or combination of networks.

**[0147]** The address of a target Client 12 needs to be known so that the underlying network delivers the Vox packet 95 to the correct location. With IPv4 networks, the address is typically an IPv4 Address, which is a 32-bit number that uniquely identifies a host within the network. For other networking technologies, the address could be some other type of identifier. IP networks use the Domain Name System (DNS) to resolve human-readable names into IP addresses, and the Address Resolution Protocol (ARP) to resolve IP addresses into physical addresses. Regardless of the underlying networking technology, the system 10 uses one of the above-mentioned or other known addressing schemes for delivery of Vox packets 95 to the correct location.

**[0148]** As with almost any packet-based communication system, transmitted Vox packets might not be delivered to their addressed location if the underlying network is unable to deliver the packets in which the Vox packets are encapsulated. Most packet networks do not inform transmitters when packets are dropped. Instead they rely on the transmitters and receivers to notice and compensate for any dropped packets. The system 10 is designed to use these receiver receipt report messages to coordinate this packet loss management. If the underlying network is able to inform the sender of lost or dropped packets, the system 10 utilizes this information in its retransmission protocol.

**[0149]** The management of MTU is the determination of the Maximum Transmission Unit (i.e., the maximum size of a single message) that may be sent across a network. For packet-based networks, the underlying network imposes the MTU. For circuit-switched networks, the MTU may be a tunable parameter for network efficiency and performance. Thus in most cases, the underlying network imposes or determines the maximum size of the Vox packet 95 that may be transmitted efficiently. For example with IP networks, packets may be fragmented if the payload exceeds the MTU, but at a

substantial performance penalty. With IP over Ethernet networks, the transmitting device has an MTU of 1518 bytes, as enforced by Ethernet. The largest IP packet must leave room for the Ethernet headers. The largest UDP packet must leave room for both IP and Ethernet headers and the largest Vox protocol that may be generated on Ethernet for example is the Ethernet MTU (1518) – IP header (20) – UDP header (8) = 1490 bytes. Since the Vox protocol will have a header of its own, the actual Vox media payload will be less than 1490 bytes on an Ethernet network. For Gigabit Ethernet, the MTU could be much larger, but would be determined using a similar formula.

**[0150]** In a purely packet-based network, there are two potential values for MTU, the local link MTU and the path MTU. Determining the local link MTU yields the maximum size for Vox packets to be efficiently sent out to the local network interface. The path MTU yields the maximum size of Vox packet that may be sent intact all the way to the remote node. If a sender is connected via Ethernet, the Vox packet might pass through various other systems with smaller MTUs en-route to the recipient. The smallest MTU on the path to the destination needs to be resolved and known by the sender. In the IP world, there is a standard procedure for discovering the smallest MTU called "Path MTU Discovery". For other kinds of networks, an equivalent procedure may be used. Again, since the system 10 is layered on top of other networks, any of the above MTU algorithms may be used.

## H. Operation Flow Diagrams

### H.1 Store and Stream

**[0151]** Referring to Figures 8A through 8F, a series of flow diagrams are provided to illustrate the operation of the store and stream module 24 and 84 on Clients 12 and Servers 16 respectively. Figure 8A shows the sequence of operation for a first Client $12_1$ transmitting Messages to a second Client $12_2$. Figures 8B and 8C illustrate the operation of the PIMB writer 28 and PIMB Reader 28 on the transmitting Client $12_1$. Figures 8D and 8E illustrate the operation of the PIMB Writer 28 and PIMB Reader 26 on the receiving Client $12_2$. Figure ~~10F~~ **8F** illustrates a flow diagram of the Store and Steam module 84 on a server 16.

**[0152]** In Figure 8A, the User of Client $12_1$ running on a Device $13_1$ originates Media to be transmitted. The Media can be originated at the Device 13 in a number of different ways such the User creating Media by speaking into a microphone or creating video

content on their Device 13. Media is also originated by a Device 13 by receiving sensor data, such a GPS information or a temperature reading. Regardless of how the Media is originated, the Media is encoded by the PIMB Writer 28 (box 130), which converts the Media into indexed media payloads and stores them in the PIMB 30 (box 132) on Client $12_1$. The PIMB Reader 26 on the Client $12_1$ reads the payloads out of the PIMB 30, creates Vox packets, and transmits the packets to the receiving Client $12_2$ (box 134) over the network 18. Each Server 16 along the path between the sending Client $12_1$ and the receiving Client $12_2$ stores the transmitted payloads in the PIMB 85 and transmits the Vox packets to the next hop (box 133). At the receiving Client $12_2$, the net receive function of the PIMB Writer 28 converts the Vox packets into indexed media payloads (box 136) and stores the payloads into the PIMB 30 of Client $12_2$ (box 138). The rendering module of the PIMB reader 26 on Client $12_2$ renders the payload information read from the PIMB 30 into a medium suitable for human consumption, such as voice or video (box 140). Each of these steps are described in more detail below with respect to Figures 10B through 10E.

**[0153]** In Figure 8B, a sequence of the Encode Receive function performed by the PIMB Writer 28 (step 130 of Figure 8A) is provided in detail. In the initial step $130_1$, the User of the Device 13 running the Client $12_1$ originates the Media to be transmitted. As noted above, the Media may be derived by speaking into a microphone, using a camera, receiving sensor data, or by some other Media generating component. In the next step $130_2$, the Encode Receiver 28a encodes the Media and creates the indexed media payloads (step $130_3$), which are then stored in the PIMB 30 (step 132) by the Data storer 28c.

**[0154]** In Figure 8C, the sequence of the Transmit function performed by the PIMB Reader 26 (step 134 of Figure 8A) on the sending client $12_1$ is provided in detail. In decision loop $134_1$, the transmit function of the PIMB Reader 26 continuously checks to see if indexed media payloads that are to be transmitted have been written into the PIMB 30 and are available for transmission. If such payloads are available in the PIMB 30, the Data Prioritizer 26f prioritizes the payloads that should be sent first, using the MCMS Participant Attributes information, as illustrated in step $134_2$. Information about the highest priority payloads are passed to the Packet Optimizer module 26g which runs the PQM (step $134_3$), as described in more detail below with respect to Figures 9A-9C. The appropriate payloads are then retrieved from the PIMB 30 (step $134_4$) by the Data

Retriever 26h and converted into Vox packets 95 by the Packetizer 26i (step $134_5$). The Vox packets 95 are then transmitted (step $134_6$) by the Transmitter 26j over the network 18 to the receive Client $12_2$, which sends back receipt reports reflecting the properties (loss, jitter, throughput) of the packets that have been received. These receipt reports provide the information necessary for the PQM to calculate the MABR for a given receiver. The aforementioned process is repeated for each transmission loop as indicated by the return arrow from the transmit step to the top of the flow chart.

**[0155]** In the embodiment described above, the media is encoded, stored in the PIMB 30 and then transmitted over the network in a serial fashion. In an alternative embodiment, the encoded media can be stored in the PIMB 30 and transmitted over the network in parallel, meaning the two functions occur substantially at the same time.

**[0156]** In Figure 8D, the sequence for the Net Receive function (step 136 of Figure 8A) of the PIMB Writer 28 on the receiving Client $12_2$ is illustrated. In the initial step $136_1$, the network receiver 28d receives the Vox packets 95 over the network 18. The Data Storer 28f converts the packets into indexed media payloads (step $136_3$), which are stored in the PIMB 30 (step $136_4$). As the payloads are stored, the Data Quality Manager (DQM) 28g is run. The DQM 28g checks for missing or corrupted packets, ensures the eventually storage of an exact copy of the transmitted data, and the sends receipt reports regarding the conditions of the network to the transmitter. Each of these functions of the DQM 28g are described in more detail below with regard to Figures 9D through 9F.

**[0157]** In Figure 8E, the sequence for the Render function of the PIMB Reader 26 (box 140 of Figure 8A) on the receive Client $12_2$ is illustrated. In the initial step $140_1$, the Data Prioritizer 26a prioritizes the indexed media payloads to be rendered as determined by the MCMS application 20 using the MTSD information as well as User status and presence information, including the User's Intentions and Attention status. The prioritized payloads are then read from the PIMB 30 (step $140_2$) by the Data Retriever 26b. The PLC/Interpolator 26c performs packet loss compensation and interpolation (step $140_3$) on the retrieved payloads, using known packet loss compensation and interpolation algorithms depending on which Codecs are used. In the next step, the Mixer 26d mixes (step $140_4$) multiple Messages of a Conversation if two or more Participants have generated Media at the same time within the same Conversation (e.g., both are speaking at the same time). The Renderer 26e renders (step $140_5$) the data

stream from the Mixer 26d, generating sound, video, or other Media (step $140_6$) for the recipient User.

**[0158]** In Figure 8F, the sequence for a Server 16 to receive Vox packets from the previous hop on the network 18, store, archive and transmit the Vox packets to the next hop is illustrated. In the initial step, the Server 16 performs the Net Receive function of the PIMB writer (similar to Fig. 8D) to store the indexed media payloads of the received data in the PIMB 85 and archive 89 or the Server 16. The server 16 also performs the Transmit function of the PIMB writer (similar to Fig. 8C) to forward the received packets onto the next hop on the network 18. In this manner, a copy of the media generated by the transmit Client $12_1$ is received, stored and transmitted at each hop along the path to the receive Client $12_2$.

**[0159]** In the aforementioned embodiment, the writing of received indexed media is stored in the PIMB 91 of the Server 16 and transmitted to the next hop in a serial fashion. In an alternative embodiment, the received indexed media payloads can be stored in the PIMB 91 and transmitted to the next hop substantially at the same time. The storage of Media on the PIMB 30 of both transmitting and receiving Devices 13 allows for the progressive transmission and rendering of Media. On the transmit side, the Media originating on the transmitting device may be progressively transmitted over the network as it is being received. In various embodiments, the encoded Media (regardless of how it is originated) may be progressively transmitted before, after, or at substantially the same time it is stored in the PIMB 30. On the receive side, the incoming Media may also be progressively rendered as it is received over the network, provided the User has opted to Review the Media in the near real-time mode. In various embodiments, the incoming Media may be progressively rendered before, after or substantially at the same time as it is stored in the PIMB 30 of the receiving Device 13. If the received Media is to be Reviewed in the time-shifted mode, then the Media is retrieved from the PIMB 30 (or possibly a PIMB 85 on a Server 16 if replaced on the local PIMB 30) for later Review at a time designated by the User.

**[0160]** In the context of the present application, the term progressive or progressively is intended to be broadly construed and generally mean the continuous processing of a data stream based on availability of the data. For example, as Media is created or otherwise originated on a Device 13, the progressive encoding, storage, packetization and transmitting of that media is continuous, so long as the Media is available. As a

person speaks, that Media is progressive or continuously encoded, stored, packetized and transmitted for the duration of the persons speech. When the person pauses or stops speaking, there is no media to progressively process. When the person resumes speaking again, the progressive processing of the Media resumes. On the receive side, the Media is also progressive processed as the Media is being received (i.e., available). As ~~Media~~ the Media is received it is continuously stored. It will also be continually rendered as the Media is being received when in the near real-time mode or from storage when in the time-shifted mode. Although the above explanation was provided in the context of voice, it should be understood that all types of Media can be progressively processed in a similar manner. Also the progressive processing of Media does not necessarily have to be progressively processed in time-indexed order. Rather the Media is processed in the order in which it is received. If Media is received out of index order, in one embodiment, the Media is progressively processed in the order it was received and then organized into the indexed sequence in the PIMB 30. In an alternative embodiment, the received Media can be organized into its indexed sequence and then progressively rendered.

### H.2 PQM Operation Flow Diagrams

**[0161]** The PQM 26g relies on a metric called the Maximum Available Bit Rate (MABR), which is a continually computed approximation of actual transmission capacity or bandwidth (i.e., a measure of the capability of the network at a given point in time) between a sending and receiving node pair. As instantaneous network conditions change, the MABR is updated. Regular measurements of network throughput, packet loss, and jitter are considered in computing the MABR. In an alternative embodiment, the MABR may also be manually set or limited based on time of day, type of network, other conditions or parameters.

**[0162]** The PQM also considers the Intention of the recipient(s) to optimize transmission for time-sensitivity. A transmission is considered time-sensitive if either (i) the Intention of the recipient(s) is to Review the transmission "live" or in the near real-time mode, or (ii) the recipient would like to immediately Review a Message that for some reason is not currently stored on their Device 13 (e.g., the Message was previously stored in the archive 89). The Intention of the recipient can be either inferred by the behavior of the recipient or the recipient may set or otherwise designate their Intention. On the other hand, a transmission is considered to be not time-sensitive if the

Intention of the recipient is to Review the Message in the time-shifted mode. The Intention of the recipient to Review the transmission in either the live or time-shifted mode at least partially defines the "timeliness requirements" of the transmission. In various other embodiments, factors such as the urgency or priority of transmissions may also be considered in defining the timeliness requirement of the transmission.

**[0163]** The nodes in the network path between a sender and a receiving pair also need to be consistent regarding the status of intentions of the recipients. If one target recipient indicates timeliness, meaning they wish to Review the transmission immediately or live, then all the intermediate nodes on the network along the sender-receiver path need to have the same timeliness requirement, regardless of the requirements of other recipients. The timeliness requirement of each of the intermediate nodes is therefore dependent on the receiving nodes the transmission is being sent to. This dependency is sometimes referred to as a "union of requirements" for target nodes in the network transmission path.

**[0164]** The PQM further considers an Ideal Bit Rate or "IBR" for each scheduled Message payload transmission. For time-sensitive transmissions, the IBR is computed based on the packetization rate needed for substantially real time or live communication (referred to herein as the Real Time Bit Rate or RTBR). With voice for example, a packetization rate of a packet every 20 milliseconds containing 20 milliseconds of audio data is considered an acceptable IBR for conducting live conversations. The RTBR for such a system in kilobits per second would be the size of 1 second of audio payload data plus the size of all network headers that would be generated for the transmission. For video media or a combination of voice and video, the RTBR will likely be substantially higher than simply voice. For other types of media such as sensor or GPS positioning data, the RTBR will likely be lower than that of voice. For non time-sensitive transmissions, the IBR is set to a Maximum Efficiency Bit Rate (MEBR) to optimize the usage or efficiency of transmissions over the network. The MEBR is calculated by adjusting the packetization rate to its largest possible value for the underlying network. If multiple messages or payloads are to be sent between a sending and receiving pair, then an Aggregate IBR (AIBR) is considered for the transmission.

**[0165]** The PQM operates by sending data in a series of transmission loops for each sending and receiving pair. The transmission loops for each sending and receiving pair are independent. Any transmission on the network may affect the MABR of other

sending-receiving pairs. Accordingly, the MABR is preferably continually computed for all recipients.

**[0166]** Referring to Figures 9A through 9C, flow charts illustrating the sequence of operation of the PQM for a single sending and receiving pair is illustrated. In Figure 9A, the steps in determining the MABR between the single sending and receiving pair are illustrated. In Figure 9B, a flow chart illustrating the steps for calculating the AIBR for each transmission loop for the single sending and receiving pair are illustrated. In Figure 9C, a sequence for determining the amount of data to transmit between the sending and receiving pair per loop is illustrated. The processes illustrated in the three diagrams run simultaneously and interact with one another, as described in more detail below.

**[0167]** In Figure 9A, a flow chart 50 for calculating the MABR for the network interface between the sending and receiving pair is shown. In the initial step $50_1$, the network interface between the sending and receiving pair is monitored. The sender periodically receives reports, which contain information regarding the status of the network connection at the receiver in step $50_2$. The reports include information regarding the current status of data throughput $50_3$, packet loss $50_4$, and jitter $50_5$ as observed by the receiver at the network interface. In step $50_6$, the MABR is calculated at the sender based on these observations contained in the reports. By monitoring or observing the data in these reports, the MABR value is continually adjusted based on current network capabilities or conditions between the sending and receiving pair. As network capabilities become more favorable for data transmission, the MABR will increase. If the network capabilities become less favorable for transmission, the MABR will decrease, potentially all the way to zero for an unusable network. The reports are similar to the packet loss reports generated by nodes in TCP networks, but additionally include throughput and jitter information as well.

**[0168]** If there are several network interfaces between the sending and receiving pair, an MABR is calculated for each interface for which a receipt report is received. If no traffic has been recently sent on the network, or no receipt reports have been received, the MABR may not reflect current network conditions. However, since receipt reports are generated continually by receivers while data is transmitted, the sender's MABR metrics will quickly converge to a more accurate value.

**[0169]** Referring to Figure 9B, a flow chart 52 illustrating the steps for calculating the AIBR for a transmission loop is illustrated. In the initial step $52_1$, the Messages with Media (by which we mean portions of the time indexed media that belongs to this Message) ready to be transmitted between the sending and receiving pair in the current loop are ascertained. A list of Messages with Media is then built $52_2$. For each Message in the list $52_3$, the time-sensitivity or timeliness requirement of each Message is considered $52_4$. If a particular Message is not time-sensitive, then the IBR is set to the Maximum Efficiency Bit Rate (MEBR) $52_5$. On the other hand, if a Message is time-sensitive, then the IBR is set to the Real Time Bit Rate (RTBR) $52_6$. In the next step $52_7$, the previously computed IBRs for each of the Messages in the list are summed together, resulting in the Aggregate Ideal Bit Rate (AIBR) $52_8$ for the transmission loop. As signified by the return arrow $52_9$, the above-described process is repeated for each transmission loop between the sending and receiving pair.

**[0170]** Referring to Figure 9C, a flow chart 54 illustrating the sequence for determining the rate of data to transmit between the sending and receiving pair per transmission loop is illustrated. In the initial step $54_1$, the MABR (as calculated in Figure 9A) is compared to the AIBR (as determined in Figure 9B) for the next transmission.

**[0171]** If the MABR is greater than or equal to the AIBR, then all the Messages identified as ready for transmission in the loop are packetized at the IBR rate $54_2$ and transmitted $54_3$.

**[0172]** On the other hand, if the MABR is less than the AIBR, then a series of procedures are applied to so that the PQM meets its goals of the timely delivery of an adequate copy of the data, the efficient use of available bandwidth, and/or adjustments to the payload content and quality, packet size, and transmission rate to meet current network conditions.

**[0173]** In the initial step, the Messages in the list are reviewed for time sensitivity $54_4$. If there are no time sensitive Messages, then the bit rate is reduced to the MABR $54_5$, and the Messages are transmitted $54_3$.

**[0174]** If the list includes time-sensitive Messages, the bit rate allocated for the non time-sensitive Messages is reduced $54_6$ until it meets the MABR limits. If reducing the bit rate all the way to zero is insufficient to meet the MABR, then these non time-sensitive Messages are removed from the list of Messages to be transmitted in the loop.