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(54) Title: METHOD AND SYSTEM FOR PROVIDING CONTENTION-FREE CHANNEL ACCESS SERVICE IN A COMMUNICATION NETWORK

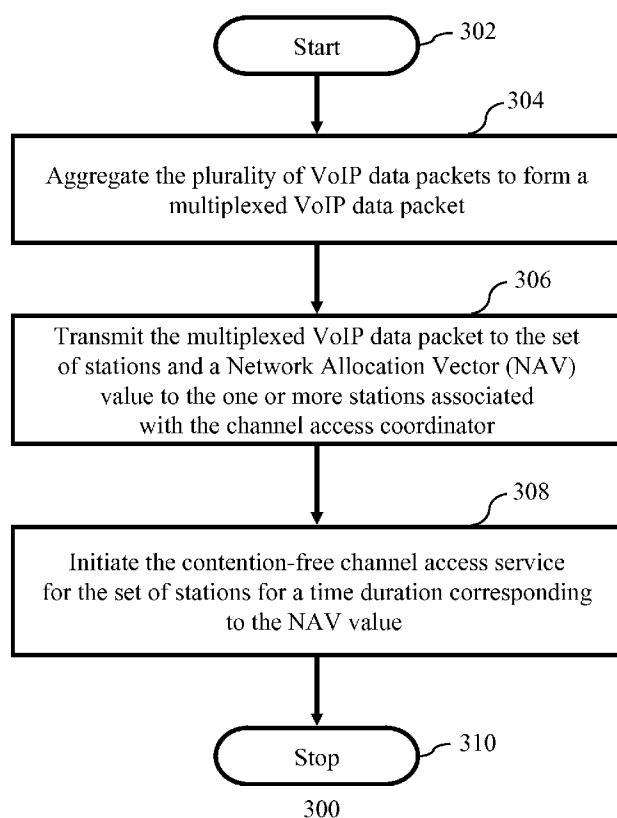


FIG. 3

(57) Abstract: A method (300) and Base Station (BS) for providing contention-free channel access service is provided. The method includes aggregating (304) a plurality of VoIP data packets to form a multiplexed VoIP data packet. The multiplexed VoIP data packet is associated with a set of stations and each station of the set of stations is selected from the one or more stations associated with the channel access coordinator. Further, the method includes transmitting (306) the multiplexed VoIP data packet to the set of stations and a Network Allocation Vector (NAV) value to the one or more stations associated with the channel access coordinator. The NAV value is determined based on at least one of the set of stations and the multiplexed VoIP data packet. Furthermore, the method includes initiating (308) the contention-free channel access service for the set of stations for a time duration corresponding to the NAV value.



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METHOD AND SYSTEM FOR PROVIDING CONTENTION-FREE CHANNEL ACCESS SERVICE IN A COMMUNICATION NETWORK

FIELD OF THE INVENTION

[0001] The present invention relates generally to the field of communication networks, and more specifically, to a method and system for providing contention-free channel access service in a communication network.

BACKGROUND OF THE INVENTION

[0002] With the advent of technology, communication methodologies are evolving each day. Along with the conventional wired and wireless communication methodology, Voice over Internet Protocol (VoIP), has gained popularity in today's world. Conventionally, communication by using VoIP methodology is done through transmission of VoIP data packets between one or more base stations (BSs) in a communication network. Typically, a VoIP data packet can include audio data. Further, each data packet starts with a header. The header can include information regarding the type of data carried by the packet, the sender's information, the receiver's information etc. The header can form an appreciable part of the size of the data packet. Further, client devices such as mobile phones, computers and stations are associated with these BSs. The BSs facilitate VoIP communication between the client devices over a communication network. As an example of the VoIP communication methodology, consider a case where a first station connected to a first BS, initiates a VoIP communication with a second station connected to a second BS. The first station can send an up-link VoIP data packet to the first BS. The first BS transmits this data packet to the second BS. Further, the second BS sends the packet to the second station as a down-link data packet, thus completing the process of VoIP data packet transmission. The second station can reply to the received VoIP data packet by sending an up-link data packet

to the second BS. The second BS can transmit the data packet to the first BS, which then sends the packet to the first station. In this case, the first and the second BSs act as access points in the network. Further, this process of transmission can be continued till the VoIP communication is terminated.

However, the above mentioned method of VoIP communication suffers from some inherent inefficiency. Such inefficiencies can be, for example, inefficiency in accessing a channel for transmission of data packets, large header size of the data packets, different transmission rates for different data packets, and, in some cases, different transmission rates for different parts of the same data packet. For example, the header part of the data packet may be transmitted at a different transmission rate than the data part of the same data packet. Consequently, due to these inherent inefficiencies, the practical number of possible a VoIP session decreases considerably. Moreover, in the presence of background traffic, the VoIP capacity further reduces. As a result, the channel for VoIP communication remains under - utilized.

BRIEF DESCRIPTION OF THE FIGURES

[0003] The accompanying figures, where like reference numerals refer to identical or functionally similar elements throughout the separate views, and which, together with the detailed description below, are incorporated in and form part of the specification, serve to further illustrate various embodiments and explain various principles and advantages, all in accordance with the present invention.

[0004] **FIG. 1** illustrates an exemplary communication network, where various embodiments of the present invention can be practiced;

[0005] **FIG. 2** illustrates a block diagram of an exemplary Base Station (BS), in accordance with various embodiments of the present invention;

[0006] **FIG. 3** depicts a flow diagram illustrating a method for providing contention-free channel access service, in accordance with an embodiment of the present invention;

[0007] **FIGs 4 and 5** depict a flow diagram illustrating a method for providing contention-free channel access service, in accordance with another embodiment of the present invention; and

[0008] **FIG. 6** depicts a communication flow diagram illustrating a method for providing contention-free channel access service, in accordance with various embodiments of the present invention.

[0009] Skilled artisans will appreciate that elements in the figures are illustrated for simplicity and clarity and have not necessarily been drawn to scale. For example, the dimensions of some of the elements in the figures may be exaggerated, relative to other elements, to help in improving an understanding of the embodiments of the present invention.

DETAILED DESCRIPTION

[0010] Before describing in detail the particular method and system for providing contention-free channel access service in a communication network, in accordance with various embodiments of the present invention, it should be observed that the present invention resides primarily in combinations of method steps and apparatus components related to providing contention-free channel access service in a communication network. Accordingly, the apparatus components and method steps have been represented, where

appropriate, by conventional symbols in the drawings, showing only those specific details that are pertinent for an understanding of the present invention, so as not to obscure the disclosure with details that will be readily apparent to those with ordinary skill in the art, having the benefit of the description herein.

[0011] In this document, the terms ‘comprises’, ‘comprising’ or any other variation thereof, are intended to cover a non-exclusive inclusion, such that a process, method, article or apparatus that comprises a list of elements does not include only those elements but may include other elements that are not expressly listed or inherent in such a process, method, article or apparatus. An element preceded by ‘comprises ... a’ does not, without more constraints, preclude the existence of additional identical elements in the process, method, article or apparatus that comprises the element. The term ‘another’, as used in this document, is defined as at least a second or more. The terms ‘includes’ and/or ‘having’, as used herein, are defined as comprising.

[0012] In accordance with various embodiments of the present invention, a method for providing contention-free channel access service for a plurality of Voice over Internet Protocol (VoIP) data packets in a communication network is provided. The plurality of VoIP data packets can be received at a channel access coordinator of the communication network. Further, one or more stations of the communication network can be associated with the channel access coordinator. The method includes aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet. The multiplexed VoIP data packet is associated with a set of stations and each station of the set of stations is selected from the one or more stations associated with the channel access coordinator. Further, the method includes transmitting the multiplexed VoIP data packet to the set of stations and a Network Allocation Vector (NAV) value to the one or more stations associated with the channel

access coordinator. The NAV value is determined based on at least one of the set of stations and the multiplexed VoIP data packet. Furthermore, the method includes initiating the contention-free channel access service for the set of stations for a time duration corresponding to the NAV value.

[0013] In accordance with various embodiments of the present invention, a Base Station (BS) for providing contention-free channel access service to a plurality of VoIP data packets in a communication network is provided. The plurality of VoIP data packets can be received at the BS. Further, one or more client-devices of the communication network can be associated with the BS. The BS includes an aggregator for aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet. The multiplexed VoIP data packet is associated with a set of client-devices and each client-device of the set of client-devices is selected from the one or more client-devices associated with the BS. Further, the BS includes a transceiver configured to transmit the multiplexed VoIP data packet to the set of client-devices and a Network Allocation Vector (NAV) value to the one or more client-devices associated with the BS. The NAV value is determined based on at least one of the set of client-devices and the multiplexed VoIP data packet. Furthermore, the BS includes a processor configured to initiate the contention-free channel access service for the set of client-devices for a time duration corresponding to the NAV value.

[0014] **FIG. 1** illustrates an exemplary communication network **100**, where various embodiments of the present invention can be practiced. The communication network **100** can be, for example, an 802.11 compatible network. The communication network **100** includes a first channel access coordinator **102** and a second channel access coordinator **104**. The first and the second channel access coordinators **102** and **104** can be, for example, Base Stations (BSs) of the communication network **100**. Further, one or more stations of

the communication network **100** can be associated with the first channel access coordinator **102** and the second channel access coordinator **104**. These stations can be, for example, client devices, communication devices, mobile phones, Personal Digital Assistants (PDAs), computers and/or laptops. As depicted in **FIG. 1**, stations **106**, **108**, **110**, **112** and **114** are associated with the first channel access coordinator **102**. Further, stations **116**, **118**, **120**, and **122** are associated with the second channel access coordinator **104**. These stations can communicate with each other with the help of the first and the second channel access coordinator **102** and **104**. For one embodiment, the communication can be a Voice over Internet Protocol (VoIP) based communication.

[0015] Typically, the channel access coordinators transmit data over a particular communication channel to facilitate communication between different stations. The communication channel can have a predefined channel bandwidth, for example, 20 Mega Hertz (MHz) to 40 MHz. For one embodiment, the first access coordinator **102** and the second access coordinator **104** may utilize a particular channel to facilitate communication between the stations **106**, **108**, **110**, **112**, **114** and the stations **116**, **118**, **120**, **122**. Before a communication can take place the channel has to be sensed for availability. The communication can take place only when the channel is available for transmitting data. Typically, when one or more stations sense the channel for availability before transmitting their respective data packets, the stations can be assumed to ‘contend’ for the channel. Based on predefined communication protocols, whichever station finds the channel available for data transmission, transmits its data packets over the channel. Consequently, such a mode of communication can be called ‘contention-mode’ of communication. In IEEE 802.11 standards, the ‘contention-mode’ of communication is also termed as a regular Distributed Co-ordinate Function (DCF) service of communication.

[0016] However, some communication protocols do not require a station to sense the channel before transmitting their data packets. For example, a station can be ‘polled’ to determine if the station is interested in transmitting any data packets. For example, a channel access coordinator may reserve the channel and ask a particular station whether it wants to transmit any data packet. In this case, a station is not required to ‘contend’ for the channel in order to transmit data packets. Consequently, such a mode of communication can be called ‘contention-free mode’ of communication.

[0017] In the case of ‘contention-mode’ of communication or regular DCF mechanism, all the stations corresponding to a channel access coordinator sense the channel individually for availability. As an example, the stations **116**, **118**, **120** and **122** can sense the channel individually, on which the second channel access coordinator **104** is working. When the station **116** senses the channel to be available for transmission, it transmits and up-links VoIP data packet to the second channel access coordinator **104**. Further, while the station **116** is transmitting its data packets, the stations **118**, **120** and **122** would not sense the channel for availability. In other words, the channel bandwidth is dynamically allocated or ‘booked’ for the station **116** for a predefined time period. This predefined time period is called Network Allocation Vector (NAV) value. For one embodiment, the NAV value may be preset based on a communication protocol being used for communication.

[0018] Channel access in the contention mode can be better explained by using the following example. Consider a case when a first user, using the station **116**, wants to communicate with a second user, using the station **106**. The communication can be a VoIP based communication. To initiate the communication, the station **116** senses the channel for availability. When the channel is available for transmission, the station **116** sends an up-link VoIP data packet. The VoIP data packet, typically, includes a header and data

which needs to be communicated. Generally, the header includes a communication address of the sender, i.e. the station **116**, and the communication address of the station to which the packet is sent, i.e. the station **106**. The header may form a major part of the size of the VoIP data packet. As an example, consider a case when the VoIP data included in a data packet has a size of 30 bytes. As the VoIP data packet is transmitted, 40 bytes of Real-Time Transport Protocol (RTP) and User Datagram Protocol (UDP) overhead headers are added to the data packet. The RTP and UDP are communication protocols for facilitating transmission of audio or video data. Further, for the 802.11b network, 34 bytes of Media Access Control (MAC) overhead header are also added in the data packet. Like RTP and UDP, MAC also helps in transmission of data packet over the available channel. Additional physical (PHY) layer overhead headers may also be added in the data packet. Consequently, the header size becomes a considerable part of the total VoIP data packet. In the present example, the data size is 30 bytes as compared to the header size of 74 bytes (34 bytes + 40 bytes).

[0019] The VoIP data packet is transmitted in the up-link by the station **116** to the second channel access coordinator **104**. After receiving the VoIP data packet, the second channel access coordinator **104** can transmit the packet to the first channel access coordinator **102**. For one embodiment, before transmitting, the second channel access coordinator **104** may sense the channel for availability, on which the first channel access coordinator **102** and the second channel access coordinator **104** are working. When the channel is available, the VoIP data packet can be transmitted by the second channel access coordinator **104** to the first channel access coordinator **102**. For another embodiment, the first channel access coordinator **102** and the second channel access coordinator **104** can be connected by a wired backhaul, such as Ethernet. In this case, the second channel access coordinator **104** does

not have to sense the wireless channel before transmitting the VoIP data packet to the first channel access coordinator **102**.

[0020] After receiving the VoIP data packet, the first channel access coordinator **102** transmits the packet to the station **106**. For one embodiment, before transmitting the VoIP data packet to the station **106**, the first channel access coordinator **102** senses the channel for availability. Further, if the second user, using the station **106**, wants to reply to the received VoIP data packet, the station **106** would have to sense the channel corresponding to the first channel access coordinator **102** for availability. This process is same as discussed for the sensing of the channel by the station **116**.

[0021] Another method for transmitting data packets over the network can be the 'contention-free mode' of communication. In this method, the channel access coordinator polls its corresponding stations to check whether any of the stations want to transmit an up-link VoIP data packet. As an example, the second channel access coordinator **104** can poll the stations **116**, **118**, **120** and **122** to check whether these stations want to up-link a VoIP data packet. For one embodiment, the polling of the stations can be performed in a predefined sequence. For example, the second channel access coordinator **104** can poll the station **116** first, then the stations **118**, **120** and **122**, in that order. For another embodiment, a channel access coordinator can poll the respective stations based on their data transmission requirements or a previous request for data transmission. Further, during the polling of the station **116**, the second channel access coordinator **104** can send a poll signal to the station **116** and wait for an up-link data packet from the station **116**. Further, an NAV value can be set for the duration of polling. The NAV value may be preset based on the communication protocol being used.

[0022] During the NAV time duration, the stations **118**, **120** and **122** do not sense the channel and thus they cannot up-link any VoIP data packet. On being polled by the second channel access coordinator **104**, the station **116** can up-link a VoIP data packet during the NAV time duration. If no packet is received by the second channel access coordinator **104** from the station **116** after waiting for a predefined time period, the next station, e.g. the station **118** can be polled.

[0023] A VoIP communication by using 'contention-free mode' can be understood by the following example. Consider a case when a first user using the station **116** wants to communicate with a second user, using the station **106**. The first user can send a data packet to the second access coordinator **104** by using the station **116** when the station **116** is polled. This data packet can be sent in a form of a VoIP data packet. As described earlier, the VoIP data packet also includes a header along with the data. After receiving the VoIP data packet, the second channel access coordinator **104** may sense the channel for availability, on which the first channel access coordinator **102** is working. For one embodiment, the first channel access coordinator **102** and the second channel access coordinator **104** can be connected by a wired overhauled and thus the sensing of the wireless channel is not required. Thereafter, the VoIP data packet can be transmitted to the first channel access coordinator **102**. After receiving the VoIP data packet, the first channel access coordinator **102** sends the packet to the station **106**.

[0024] After the VoIP data packet is received by the station **106**, it is expected that the second user will respond to the received data packet. The first channel access coordinator **102** sends a poll signal to the station **106** to indicate that a response is expected from the station **106**. The poll signal lasts for a predefined time duration and the station **106** can respond within this time duration. Further, while the VoIP data packet is being down-

linked to the station **106**, an NAV value is set by the first channel access coordinator **102**. During the NAV time duration none of the other stations, i.e. the stations **108**, **110**, **112** and **114** use the channel for communication. In other words, the channel is booked for the first channel access coordinator **102** during the NAV time duration and it uses the channel to poll and receive a reply from the station **106**. The stations **108**, **110**, **112** and **114** do not sense the channel during the NAV time period. These stations wait for the NAV time duration to lapse before they sense the channel again. After the NAV time duration is lapsed, the channel may be available for other stations to contend for availability.

[0025] The second user, using the station **106**, can respond by sending a data packet, after receiving the VoIP data packet from the station **116**. The response can be sent during the NAV time duration. The first access coordinator **102** transmits the VoIP data packet received from the station **106** to the second channel access coordinator **104**, which in turns sends the packet to the station **116**. Further, there may also be a case when the second user wants to end the communication with the first user. In this case, the station **106** does not up-link any VoIP data packet to the first access coordinator **102**, thus indicating the end of the conversation.

[0026] **FIG. 2** illustrates a block diagram of an exemplary Base Station (BS) **200**, in accordance with various embodiments of the present invention. One or more stations of the communication network **100** can be associated with the BS **200**. Examples of the stations can be client devices, mobile phones, Personal Digital Assistants (PDAs), laptops, and computers. For one embodiment, the BS **200** can receive a plurality of data packets transmitted from another BS of the communication network **100**. The data packets can be, for example, VoIP data packets. Each of these VoIP data packets start with a header, which

can include, for example, information about the VoIP data packet, the source of the VoIP data packet and the destination of the VoIP data packet.

[0027] The BS **200** can include an aggregator **202**, a transceiver **204** and a processor **206**. The aggregator **202** can be configured to aggregate the plurality of VoIP data packets to form a multiplexed VoIP data packet. For one embodiment, the aggregator **202** can be configured to aggregate the VoIP data packets received at the BS **200** over a predefined period of time. The multiplexed VoIP data packet is associated with a set of client-devices. The set of client-devices includes those client-devices of the communication network **100** which are destinations for any of the plurality of VoIP data packets received at the BS **200**. For example, consider a case when a plurality of stations has sent one or more VoIP data packets to the BS **200**. The aggregator **202** aggregates all the VoIP data packets collected over a predefined time interval and forms a multiplexed VoIP data packet. This single multiplexed VoIP data packet contains all the data packets received during the predefined time duration. Further, a common header is formed for the multiplexed VoIP data packet. The common header can include the information relating to the communication addresses of the destination stations. The multiplexed VoIP data packets can then be transmitted to the corresponding destination stations.

[0028] Further, the transceiver **204** can be configured to transmit the multiplexed VoIP data packet to the destination stations. The transceiver **204** can also transmit a Network Allocation Vector (NAV) value to the one or more stations associated with the BS **200**. For one embodiment, the NAV value can be determined based on at least one of, the destination stations and the multiplexed VoIP data packet. Further, the transceiver **204** can form a multicast group for the destination stations and multicast the multiplexed VoIP data packet. The transceiver **204** can also be configured to poll each station of the destination stations at

a predefined interval. In this case, the transceiver **204** can poll the destination stations to receive up-link data. The concept of polling in the contention-free channel access sensing method of communication has already been explained in conjunction with **FIG. 1**.

[0029] The processor **206** can be configured to initiate the contention-free channel access service for the destination stations of the multiplexed VoIP data packet for a time duration corresponding to the NAV value. For one embodiment, the processor **206** can also be configured to determine the NAV value based on the number of destination stations for the multiplexed VoIP data packet. For example, the NAV value for five destination stations can be 100 milliseconds and the NAV value for three destination stations can be sixty milliseconds. Further, the processor **206** can also be configured to reduce a header size of the VoIP data packets received at the BS **200** by using header-compression techniques known in the art. For one embodiment, the header-compression technique can be predefined by the manufacturer of the BS **200**.

[0030] The entire process of header compression and formation of a multiplexed VoIP data after the compression can be better understood by the following example. Consider a case when the BS **200** receives five VoIP data packets within a preset time interval, say 10 milliseconds. Corresponding to the five VoIP data packets there can be five destination stations. The processor **206** can set the NAV value to be 20 milliseconds. Further, the processor **206** can compress the headers of the five VoIP data packets by using various compression techniques known in the art to form a common header. Additionally, the VoIP data packets, without the headers, can be aggregated by the aggregator **202** to form a single multiplexed VoIP data packet.. Further, the multiplexed VoIP data packet can have a common header. A multicast group can be formed for these five destination stations. The transceiver **204** can then multicast the multiplexed VoIP data packet to the corresponding

five destination stations. After the transmission, the processor **206** can initiate the contention-free channel access service for the time duration of the NAV value, i.e. 20 milliseconds. During these 20 milliseconds, the transceiver **204** polls the five destination stations for an up-link VoIP data packet. The concept of polling has already been explained in conjunction with **FIG. 1** and will be described in detail in conjunction with **FIG. 6**.

[0031] **FIG. 3** depicts a flow diagram illustrating a method **300** for providing contention-free channel access service, in accordance with an embodiment of the present invention. To describe the method **300**, reference will be made to **FIGs 1** and **2**, although it is understood that the method **300** can be implemented in any other suitable environment or network as well. Moreover, the invention is not limited to the order in which the steps are listed in the method **300**.

[0032] At step **302**, the method **300** is initiated. At step **304**, a plurality of VoIP data packets are aggregated to form a multiplexed VoIP data packet. Each VoIP data packet can include a header and data, which needs to be transmitted. For example, if two users are communicating using a VoIP communication service, the data can be an audio data. Further, the header of the VoIP data packet can include information relating to the type of the data packet, a communication address of the source client-device of the packet and a communication address of the destination client-device of the packet. When a plurality of VoIP data packets is received by a base station (BS), these packets are aggregated to form a single multiplexed VoIP data packet. During the formation of the multiplexed VoIP data packet, headers of the individual data packets are compressed and a single data packet is formed by multiplexing all the data packets. Header compression has been explained in conjunction with **FIG. 2**. Further, a common header is added to the aggregated VoIP data packet to form a multiplexed VoIP data packet. Additionally, the destination client devices

of the multiplexed VoIP data packet form a multicast group. The common header can include the information relating to the communication addresses of the destination client-devices of the multicast group. Along with the formation of the multiplexed VoIP data packet, a Network Allocation Vector (NAV) value can also be determined based on the number of the destination client-devices. The concept of determining the NAV value has already been explained in conjunction with **FIGs 1 and 2**.

[0033] At step **306**, the multiplexed VoIP data packet is transmitted to the destination client-devices and the NAV value is transmitted to all the client-devices associated with the BS. Consider an example where a BS has eight stations associated with it. The BS receives four VoIP data packets corresponding to four of its associated client-devices. The received VoIP data packets are aggregated to form a single multiplexed VoIP data packet. An NAV value can also be determined based on the number of destination client-devices, in this case four destination client-devices. Let the NAV value be 16 milliseconds for the present example. After the NAV value has been determined, the multiplexed VoIP data packet is transmitted to the four destination client-devices. Further, the NAV value is transmitted to all the eight client-devices associated with the BS. As described in conjunction with **FIGs 1 and 2**, the NAV value acts as an indicator to the stations that the channel would not be accessible for the time interval corresponding to the NAV value. Consequently, the client-devices do not sense the channel for availability during the NAV time period. The concept of channel sensing and NAV value will be further described in detail in conjunction with **FIG. 6**.

[0034] At step **308**, a contention-free channel access service is initiated for the destination client-devices. The contention for the channel access lasts for a time duration corresponding to the NAV value. During the contention-free channel access service, each

destination client-device is polled for a predetermined time period to receive an up-link VoIP data packet. The contention-free channel access service is described in detail in conjunction with **FIG. 6**. At step **310**, the method **300** is terminated.

[0035] **FIGs 4 and 5** depict a flow diagram illustrating a method **400** for providing contention-free channel access service, in accordance with another embodiment of the present invention. To describe the method **400**, reference will be made to **FIGs 1 and 2**, although it is understood that the method **400** can be implemented in any other suitable environment or network. Moreover, the invention is not limited to the order in which the steps are listed in the method **400**.

[0036] At step **402**, the method **400** is initiated. At step **404**, a plurality of VoIP data packets is received at a channel access coordinator. For one embodiment, the channel access coordinator can be a base station (BS). The plurality of VoIP data packets received by the BS can be transmitted by any other BS present in the communication network **100**. For one embodiment, these packets can be received at the BS for a duration of a predefined time period. For another embodiment, the packets can be received till a predefined number of packets have been collected at the BS. For example, a BS can collect, for example, five VoIP data packets up to duration of 20 milliseconds. At step **406**, the received VoIP data packets are aggregated to form a multiplexed VoIP data packet. Continuing with the earlier example, the five VoIP data packets are aggregated to form a single multiplexed VoIP data packet. This single data packet includes all the five VoIP data packets in a compressed form. A VoIP data packet is compressed by using various header-compression techniques known in the art. Compression of the data packets has been explained in conjunction with **FIG. 2**.

[0037] At step **408**, a multicast group of destination stations is formed based on the multiplexed VoIP data packet. For one embodiment, the forming of the multicast group can be done when the plurality of VoIP data packets are received by the channel access coordinator, i.e. before the formation of the multiplexed VoIP data packet. Typically, the information relating to the destination stations is included in a header of a VoIP data packet. The information can be, for example, the communication address of the destination stations. When a plurality of VoIP data packets are aggregated to form a multiplexed VoIP data packet, the individual headers of the VoIP data packets are compressed by using header-compression techniques. The information relating to the destination stations can be extracted and a common header can be formed for the single multiplexed VoIP data packet. Further, a multicast group is formed of the destination stations. At step **410**, an NAV value is determined based on the number of destination client-devices. Continuing with the earlier example, the NAV value corresponding to the five VoIP data packets can be, for example, 20 milliseconds. The concept of determining the NAV value has been explained in conjunction with **FIGs 1 and 2**.

[0038] At step **412**, the determined NAV value is multiplexed with the multiplexed VoIP data packet in the form of a header. The common header which includes the information relating to the multicast group of destination stations is multiplexed with the NAV value to form a common header. At step **414**, a channel is acquired for access by using a regular DCF mechanism. The regular DCF mechanism has already been explained in conjunction with **FIG. 1**. The channel is acquired to transmit the multiplexed VoIP data packet and the NAV value to all the stations. Referring to **FIG. 5**, at step **502**, the multiplexed VoIP data packet and the NAV value is transmitted to all the stations associated with the base station by using the acquired channel. As an illustrative example, a BS, having 10 stations associated with it can receive four VoIP data packets corresponding to four destination

stations. A multicast group of these four stations can be formed. Further, the four VoIP data packets can be aggregated to form a multiplexed VoIP data packet. The corresponding NAV value can be determined based on these four data packets. This NAV value is multiplexed with the multiplexed VoIP data packet in a common header. The multiplexed VoIP data packet is then transmitted to all the stations associated with the BS, 10 stations in this case. Additionally, the NAV value can also be transmitted to the stations which are not the destination stations. This is done so that all the stations can set their NAV timers according to the NAV value.

[0039] Further, after the multiplexed VoIP data packets are transmitted to the stations, the destination stations extract their own VoIP data packet from the multiplexed VoIP data packet. The process of transmission is terminated when all the VoIP data packets are extracted by their respective destination stations. At step **504**, a contention-free channel access service is initiated. In a regular Distributed Co-ordinate Function (DCF) service, every base station (BS) works on a particular channel and any station, associated with the BS, has to sense the channel for availability before it up-links any VoIP data packet. This mechanism has been explained in **FIG. 1** in the form of 'contention-mode' of communication. However, in the case of contention-free channel access service, the BS station polls a station to determine if the station wants to up-link a VoIP data packet or not. In this case, the stations do not sense the channel for availability. The stations have to wait for the polling session to begin before they can up-link their VoIP data packet.

[0040] At step **506**, polling of each station of the group of destination stations is done at a predefined interval. During the polling session, a poll signal or a poll data packet is sent to each of the destination stations for a preset time duration. A destination station can send an up-link VoIP data packet, after the preset time duration. If the station does not up-link any

VoIP data packet, another destination station of the group of destination stations is polled. As an example, consider a case when four VoIP data packets, corresponding to four destination stations, are aggregated to form a multiplexed VoIP data packet. Let the four destination stations be the first, second, third and fourth destination stations **106**, **108**, **110** and **112**. After the four stations receive their corresponding VoIP data packets, polling session is initiated. A poll signal is sent to the first destination station **106** for a preset time duration, say for 2 milliseconds. After the poll signal is sent to the first destination station **106**, the base station (BS) waits for a preset time interval, called Short Inter-frame Space (SIFS) time interval, for the first station **106** to up-link a VoIP data packet. If the first destination station **106** does not up-link any data packet after the SIFS time interval, the BS polls the second destination station **108**. For an embodiment, a Priority Inter-frame Spacing (PIFS) time interval can also be incorporated. PIFS denotes the time period after which a new VoIP data packet is up-linked by a station. Typically, PIFS time interval is lesser than SIFS time interval.

[0041] Further, when no up-link data is received, the NAV value can be adjusted and the new NAV value can be communicated to all the stations associated with the BS. On the other hand, if the first destination station **106** starts to up-link a VoIP data packet after the SIFS time interval, the BS waits for the time for which the data packet is up-linked, before it polls the second destination station **108**. The process of polling and SIFS time interval is described in detail in conjunction with **FIG. 6**.

[0042] At step **508**, the multicast group of destination stations can be updated. Continuing with the earlier example, if the first destination station **106** does not up-link any VoIP data packet after the SIFS time interval, the BS moves to other destination stations for polling. Further, the BS updates the multicast group of the destination stations and the first

destination station **106** is eliminated from the group after multiple failures of sending up-link VoIP data packets during polling by the BS. A modified multicast group is formed with the second, the third and the fourth destination stations **108**, **110** and **112** as its members. Further, in this situation, the corresponding NAV value is also changed for the modified multicast group. For example, if the NAV value for the multicast group of four destination stations was 16 milliseconds, the NAV value becomes 12 milliseconds after the update.

[0043] At step **510**, a contention-free end frame is transmitted to all the stations associated with the BS after all the stations in the multicast group have been polled and/or the time duration corresponding to the NAV value has lapsed. The end frame is transmitted after all the destination stations of the multicast group are polled. The end frame is transmitted in the form of a signal or a data packet for a preset time duration. The end frame acts as an indicator to all the stations that the contention-free channel access service has been terminated and they can sense the channel for availability. In other words, the end frame indicates the beginning of the 'contention-mode' of communication or the regular DCF mechanism. At step **512**, the regular DCF mechanism is initiated. All the stations associated with the BS can sense the channel for availability when the DCF mechanism is initiated. Any station can up-link a VoIP data packet when it senses the channel to be available. The process of aggregating and polling begins again when the BS station receives a plurality of VoIP data packets. At step **514**, the method **400** is terminated.

[0044] **FIG. 6** depicts a communication flow diagram illustrating a method for providing contention-free channel access service, in accordance with various embodiments of the present invention. As discussed in **FIG. 1**, there are two methods of transmitting data over the channel. First method is the 'contention-mode' of communication or the regular DCF

mechanism. In the regular DCF mechanism, all the stations associated with a particular base station (BS) sense the channel for availability. Whenever a station senses the channel to be available, it up-links a VoIP data packet to the BS. Second method for sensing the channel is the 'contention-free mode' of communication. In this method, the BS polls every station associated with it, one by one, to check whether a station wants to up-link any VoIP data packet. The stations do not sense the channel and wait for their polling to begin before they can up-link a VoIP data packet.

[0045] Referring to the **FIG. 6**, a channel access coordinator **602** can be associated with stations **604**, **606** and **608**. Further, the BS **602** can also be associated with other stations not shown in **FIG. 6**. The channel access coordinator **602** can be an access point or a base station (BS). For the purpose of description, the channel access coordinator is assumed to be a BS. The stations **604**, **606** and **608** can be destination stations for the VoIP data packets collected at the BS **602**. Further, a multiplexed VoIP data packet **610** is formed when a plurality of VoIP data packets, received at the BS **602**, are aggregated to form a single multiplexed VoIP data packet. The concept of aggregation and formation of a multiplexed VoIP data packet has been explained in conjunction with **FIGs. 4 and 5**.

[0046] The blocks **612**, **614** and **616** depict a poll signal transmitted by the BS to the destination stations **604**, **606** and **608** respectively. The poll signal is transmitted by the BS **602** to check whether a destination station wants to up-link a VoIP data packet or not. The method of polling and up-linking of VoIP data packet can be understood with the help of the following example. Consider a case when one or more stations associated the BS **602** wants to up-link a VoIP data packet to the BS **602**. These stations sense the channel for availability and up-link the VoIP data packet whenever the channel is available. This constitutes the regular DCF mechanism and is illustrated in the **FIG. 6** by the time period

626. During the time period **626**, the ‘contention-mode’ of communication takes place and every station individually senses the channel for availability.

[0047] Now, consider a situation when the BS **602** receives three VoIP data packets to be transmitted to the destination stations **604**, **606** and **608**. The BS **602** aggregates the VoIP data packets to form a single multiplexed VoIP data packet **610**. Further, a multicast group is formed with its members as the destination stations **604**, **606** and **608**. An NAV value is determined based on the number of the destination stations, in this case three. Assuming the NAV value to be 12 milliseconds, the multiplexed VoIP data packet, along with the NAV value is transmitted to the destination stations. For one embodiment, the NAV value is also transmitted to all the stations associated with the BS **602**.

[0048] After transmitting the multiplexed VoIP data packet **610**, all the other stations associated with the BS **602** cannot sense the channel for availability. Consequently, a dynamic bandwidth allocation or a dynamic channel allocation is done for the destination stations. This arises from the fact that a station receiving a VoIP data packet is expected to respond to that packet. Therefore, the probability of up-linking any VoIP data packets increases in the case of these destination stations. Thus, a dynamic channel allocation is done for these stations in anticipation of a response. The transmission of the multiplexed VoIP data packet to the destination stations **604**, **606** and **608** is depicted by the time period **628**. During the time period **628**, the destination stations **604**, **606** and **608** extract their respective VoIP data packets from the multiplexed VoIP data packet. For one embodiment, extraction of the VoIP data packet from the multiplexed VoIP data packet can be done by decoding the multiplexed VoIP data packet. The decoding can be done at the destination stations when the multiplexed VoIP data packet is received by them.

[0049] The time periods **630**, **634**, **638**, **642**, **646**, **650** and **654** depict a Short Inter-frame Space (SIFS) time interval of the BS **602**. Typically, SIFS is incorporated to reduce the chances of collisions that might occur between any up-link and down-link data packet. Another reason for incorporating SIFS is to allow the channel access coordinator and the associated stations to switch their radios from a transmit mode to a receive mode or vice-versa. In the present case, SIFS denotes a time period that a BS waits for any destination station to respond. The response can be in the form of a signal or a data packet. The time interval **632** depicts the time interval for which the BS **602** transmits a poll signal or a poll data packet **612** to the destination station **604**. The poll data packet **612** indicates to the destination station **604** that the BS **602** expects a response to the transmitted VoIP data packet. During the time interval **634**, the BS **602** waits for the destination station **604** to respond. After the time interval **634**, the destination station **604** can up-link a VoIP data packet, depicted as the block **618** in FIG. 6. The process of up-linking takes place in the time interval **636**. For one embodiment, if no VoIP data packet is up-linked by the destination station **604** after the time interval **634**, the BS starts polling the destination station **606** and the station **604** is eliminated from the multicast group after multiple up-link transmission failures by a station after the BS poll. Consequently, the NAV can also be changed to, say, eight milliseconds from the initial value of 12 milliseconds. However, there may be a situation where the NAV value remains the same as 12 milliseconds. In this case, the stations can reset their local NAV timers only after the whole polling session is complete.

[0050] After the VoIP data packet **618** is up-linked by the destination station **604**, the BS **602** waits for the SIFS time period **638** before it sends a poll data packet **614** to the destination station **606**. The whole process of polling and up-linking of a VoIP data packet is repeated again for the destination station **606** and then the destination station **608**. After

the destination stations **606** and **608** send their VoIP data packets **618** and **622** respectively, the BS **602** waits for the SIFS time interval **654** before it transmits an end frame packet **624** to all the associated stations. The end frame **624** is transmitted within the lapsing of the NAV time duration. Further, the end frame **624** is transmitted for a time interval **656** to all the stations associated with the BS **602**. The end frame **624** acts as an indicator to all the stations that the polling is terminated and they can sense the channel for availability. After transmission of the end frame **624**, the 'contention-mode' of communication or regular DCF is initiated.

[0051] Further, the sum of the time intervals **630**, **632**, **634**, **636**, **638**, **640**, **642**, **644**, **646**, **648**, **650**, **652**, **654** and **656** equals the NAV value for the multicast group of the destination stations **604**, **606** and **608**. In other words, the contention-free channel access service lasts for the time duration of the NAV value. Additionally, when a particular destination station is eliminated from the multicast group, the time interval corresponding to its polling and up-linking of the VoIP data packet is also eliminated from the NAV value. Consequently, a new NAV value is determined corresponding to the modified multicast group and transmitted to the associated stations. For example, if the destination station **608** is eliminated from the multicast group, a new NAV value is determined corresponding to the destination stations **604** and **606**. In this case, the end frame is transmitted to all the stations before the lapse of the original NAV value.

[0052] Those ordinarily skilled in the art can appreciate that the invention can also work if a new station, associated with the BS **602**, is added to the multicast group. If a VoIP data packet corresponding to the new station is received by the BS **602** during the regular DCF mechanism, the new station can be dynamically added in the original multicast group. Additionally, a new station can also be added if it sends a service request or a traffic request

to the BS **602**. In this case, the channel bandwidth is dynamically allocated to provide access to the station sending the traffic request. Further, the original multicast group is modified to include the new station and the NAV value is changed accordingly.

[0053] Various embodiments, as described above, provide a method and a Base Station (BS) for providing contention-free channel access service in a communication network. The present invention allows a plurality of VoIP data packets to be aggregated to form a single multiplexed VoIP data packet. In this way the headers of the individual VoIP data packets are compressed and thus the total packet size is reduced. This can facilitate in incorporating more VoIP calls within a given frequency bandwidth or channel in the down-link. Further, for up-link data transmission, a group of stations can be polled. This minimizes the time spent in sensing the channel for availability. Further, for the duration of polling of the group of stations, other stations do not sense the channel for availability. This helps in minimizing any collision or data loss in the up-link. Moreover, no additional infrastructure is required for implementing the present invention. The invention can work on the same infrastructure that is used for a regular DCF mechanism.

[0054] It will be appreciated that embodiments of the invention described herein may comprise one or more conventional processors and unique stored program instructions that control the one or more processors to implement, in conjunction with certain non-processor circuits, some, most, or all of the functions of the embodiments of the invention described herein. The non-processor circuits may include, but are not limited to, a radio receiver, a radio transmitter, signal drivers, clock circuits, power source circuits, and user input devices. As such, these functions may be interpreted as steps of a method for providing contention-free channel access service in a communication network. Alternatively, some or all functions could be implemented by a state machine that has no stored program

instructions, or in one or more application specific integrated circuits (ASICs), in which each function or some combinations of certain of the functions are implemented as custom logic. Of course, a combination of these approaches could be used. Thus, methods and means for these functions have been described herein. In those situations for which functions of the embodiments of the invention can be implemented using a processor and stored program instructions, it will be appreciated that one means for implementing such functions is the media that stores the stored program instructions, be it magnetic storage or a signal conveying a file. Further, it is expected that one of ordinary skill, notwithstanding possibly significant effort and many design choices motivated by, for example, available time, current technology, and economic considerations, when guided by the concepts and principles disclosed herein will be readily capable of generating such stored program instructions and ICs with minimal experimentation.

[0055] In the foregoing specification, the invention and its benefits and advantages have been described with reference to specific embodiments. However, one with ordinary skill in the art would appreciate that various modifications and changes can be made, without departing from the scope of the present invention, as set forth in the claims below. Accordingly, the specification and the figures are to be regarded in an illustrative rather than a restrictive sense, and all such modifications are intended to be included within the scope of the present invention. The benefits, advantages, solutions to problems, and any element(s) that may cause any benefit, advantage or solution to occur or become more pronounced are not to be construed as critical, required or essential features or elements of any or all the claims. The invention is defined solely by the appended claims, including any amendments made during the pendency of this application, and all equivalents of those claims, as issued.

[0056] The Abstract of the Disclosure is provided to comply with 37 C.F.R. §1.72(b), requiring an abstract that will allow the reader to quickly ascertain the nature of the technical disclosure. It is submitted with the understanding that it will not be used to interpret or limit the scope or meaning of the claims. In addition, in the foregoing Detailed Description, it can be seen that various features are grouped together in a single embodiment for the purpose of streamlining the disclosure. This method of disclosure is not to be interpreted as reflecting an intention that the claimed embodiments require more features than are expressly recited in each claim. Rather, as the following claims reflect, inventive subject matter lies in less than all features of a single disclosed embodiment. Thus the following claims are hereby incorporated into the Detailed Description, with each claim standing on its own as a separately claimed subject matter.

WHAT IS CLAIMED IS:

1. A method for providing contention-free channel access service for a plurality of Voice over Internet Protocol (VoIP) data packets in a communication network, the plurality of VoIP data packets being received at a channel access coordinator of the communication network, one or more stations of the communication network being associated with the channel access coordinator, the method comprising:

aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet, wherein the multiplexed VoIP data packet is associated with a set of stations, and wherein each station of the set of stations is selected from the one or more stations associated with the channel access coordinator;

transmitting the multiplexed VoIP data packet to the set of stations and a Network Allocation Vector (NAV) value to the one or more stations associated with the channel access coordinator, wherein the NAV value is determined based on at least one of the set of stations and the multiplexed VoIP data packet; and

initiating the contention-free channel access service for the set of stations for a time duration corresponding to the NAV value.

2. The method as recited in claim 1 further comprising using a header-compression technique for reducing a header size, wherein the header is associated with each of the plurality of VoIP data packets.
3. The method as recited in claim 1 further comprising forming a multicast group of the set of stations for transmission of the multiplexed VoIP data packet.

4. The method as recited in claim 1 further comprising determining the NAV value based on number of stations in the set of stations.
5. The method as recited in claim 1 further comprising multiplexing the NAV value to the multiplexed VoIP data packet in form of a header.
6. The method as recited in claim 1 further comprising decoding the multiplexed VoIP data packet at a station.
7. The method as recited in claim 1, wherein initiating the contention-free channel access service comprises polling each station of the set of stations at a predefined time interval.
8. The method as recited in claim 7 further comprising transmitting a contention-free end frame after the set of stations has been polled.
9. The method as recited in claim 8 further comprising initiating a regular Distributed Coordinate Function service after the transmission of the contention-free end frame.
10. The method as recited in claim 1 further comprising transmitting a contention-free end frame within lapsing of the time duration corresponding to the NAV value.
11. The method as recited in claim 1 further comprising updating the set of stations based on an up-link data transmitted by a station of the set of stations.

12. The method as recited in claim 1 further comprising dynamically adding a station associated with the channel access coordinator to the set of stations for the contention-free channel access service.
13. The method as recited in claim 12, wherein adding the station to the set of stations comprises receiving a service request from the station.
14. The method as recited in claim 1 further comprising sensing a communication channel of the communication network to determine an availability of the communication channel for the contention-free channel access service for the set of stations.
15. The method as recited in claim 1 further comprising dynamically allocating a channel bandwidth in the communication network to the set of stations based on a traffic request of the set of stations.

16. A Base Station (BS) for providing contention-free channel access service to a plurality of Voice over Internet Protocol (VoIP) data packets in a communication network, the plurality of VoIP data packets being received at the BS, one or more client-devices of the communication network being associated with the BS, the BS comprising:

an aggregator for aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet, wherein the multiplexed VoIP data packet is associated with a set of client-devices, and wherein each client-device of the set of client-devices is selected from the one or more client-devices associated with the BS;

a transceiver configured to transmit the multiplexed VoIP data packet to the set of client-devices and a Network Allocation Vector (NAV) value to the one or more client-devices associated with the BS, wherein the NAV value is determined based on at least one of the set of client-devices and the multiplexed VoIP data packet; and

a processor, wherein the processor is configured to initiate the contention-free channel access service for the set of client-devices for a time duration corresponding to the NAV value.

17. The BS as recited in claim 16, wherein the processor is further configured to reduce a header size by using a header-compression technique, wherein the header is associated with each of the plurality of VoIP data packets.

18. The BS as recited in claim 16, wherein the processor is further configured to determine the NAV value based on number of client-devices in the set of client-devices.

19. The BS as recited in claim 16, wherein the transceiver is further configured to poll each client-device of the set of client-devices at a predefined interval.

20. The BS as recited in claim 16, wherein the communication network is an 802.11 compatible network.

AMENDED CLAIMS

received by the International Bureau on 26 August, 2008 (26.08.2008)

WHAT IS CLAIMED IS:

1. A method for providing contention-free channel access service for a plurality of Voice over Internet Protocol (VoIP) data packets in a communication network, the plurality of VoIP data packets being received at a channel access coordinator of the communication network, one or more stations of the communication network being associated with the channel access coordinator, the method comprising:

aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet, wherein the multiplexed VoIP data packet is associated with a set of stations, and wherein each station of the set of stations is selected from the one or more stations associated with the channel access coordinator;

transmitting the multiplexed VoIP data packet to the set of stations and a Network Allocation Vector (NAV) value to the one or more stations associated with the channel access coordinator, wherein the NAV value is determined based on at least one of the set of stations and the multiplexed VoIP data packet; and

initiating the contention-free channel access service for the set of stations for a time duration corresponding to the NAV value.

2. The method as recited in claim 1 further comprising using a header-compression technique for reducing a header size, wherein the header is associated with each of the plurality of VoIP data packets.
3. The method as recited in claim 1 or claim 2 further comprising forming a multicast group of the set of stations for transmission of the multiplexed VoIP data packet.
4. The method as recited in any one of claims 1 to 3 further comprising determining the NAV value based on number of stations in the set of stations.
5. The method as recited in any one of claims 1 to 4 further comprising multiplexing the NAV value to the multiplexed VoIP data packet in form of a header.
6. The method as recited in any one of claims 1 to 5 further comprising decoding the multiplexed VoIP data packet at a station.
7. The method as recited in any one of claims 1 to 6, wherein initiating the contention-free channel access service comprises polling each station of the set of stations at a predefined time interval.

8. The method as recited in any one of claims 1 to 7, further comprising transmitting a contention-free end frame within lapsing of the time duration corresponding to the NAV value.
9. The method as recited in any one of claims 1 to 8, further comprising updating the set of stations based on an up-link data transmitted by a station of the set of stations.
10. The method as recited in any one of claims 1 to 9, further comprising dynamically adding a station associated with the channel access coordinator to the set of stations for the contention-free channel access service.
11. The method as recited in any one of claims 1 to 10, further comprising sensing a communication channel of the communication network to determine an availability of the communication channel for the contention-free channel access service for the set of stations.
12. The method as recited in any one of claims 1 to 11, further comprising dynamically allocating a channel bandwidth in the communication network to the set of stations based on a traffic request of the set of stations.

13. A Base Station (BS) for providing contention-free channel access service to a plurality of Voice over Internet Protocol (VoIP) data packets in a communication network, the plurality of VoIP data packets being received at the BS, one or more client-devices of the communication network being associated with the BS, the BS comprising:

an aggregator for aggregating the plurality of VoIP data packets to form a multiplexed VoIP data packet, wherein the multiplexed VoIP data packet is associated with a set of client-devices, and wherein each client-device of the set of client-devices is selected from the one or more client-devices associated with the BS;

a transceiver configured to transmit the multiplexed VoIP data packet to the set of client-devices and a Network Allocation Vector (NAV) value to the one or more client-devices associated with the BS, wherein the NAV value is determined based on at least one of the set of client-devices and the multiplexed VoIP data packet; and

a processor, wherein the processor is configured to initiate the contention-free channel access service for the set of client-devices for a time duration corresponding to the NAV value.

14. The BS as recited in claim 13, wherein the processor is further configured to reduce a header size by using a header-compression technique, wherein the header is associated with each of the plurality of VoIP data packets.
15. The BS as recited in claim 13 or claim 14, wherein the processor is further configured to determine the NAV value based on number of client-devices in the set of client-devices.
16. The BS as recited in any one of claims 13 to 15, wherein the transceiver is further configured to poll each client-device of the set of client-devices at a predefined interval.

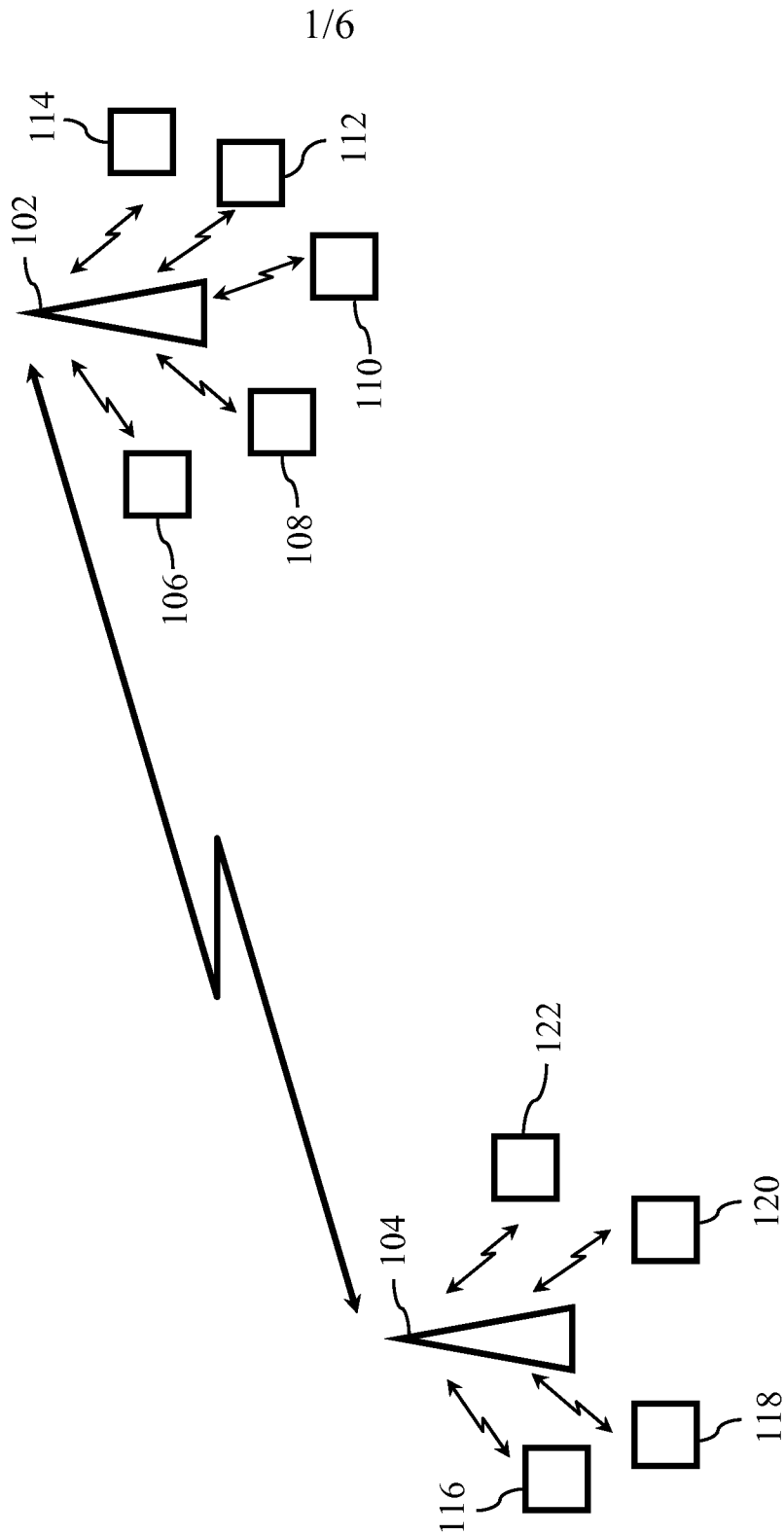
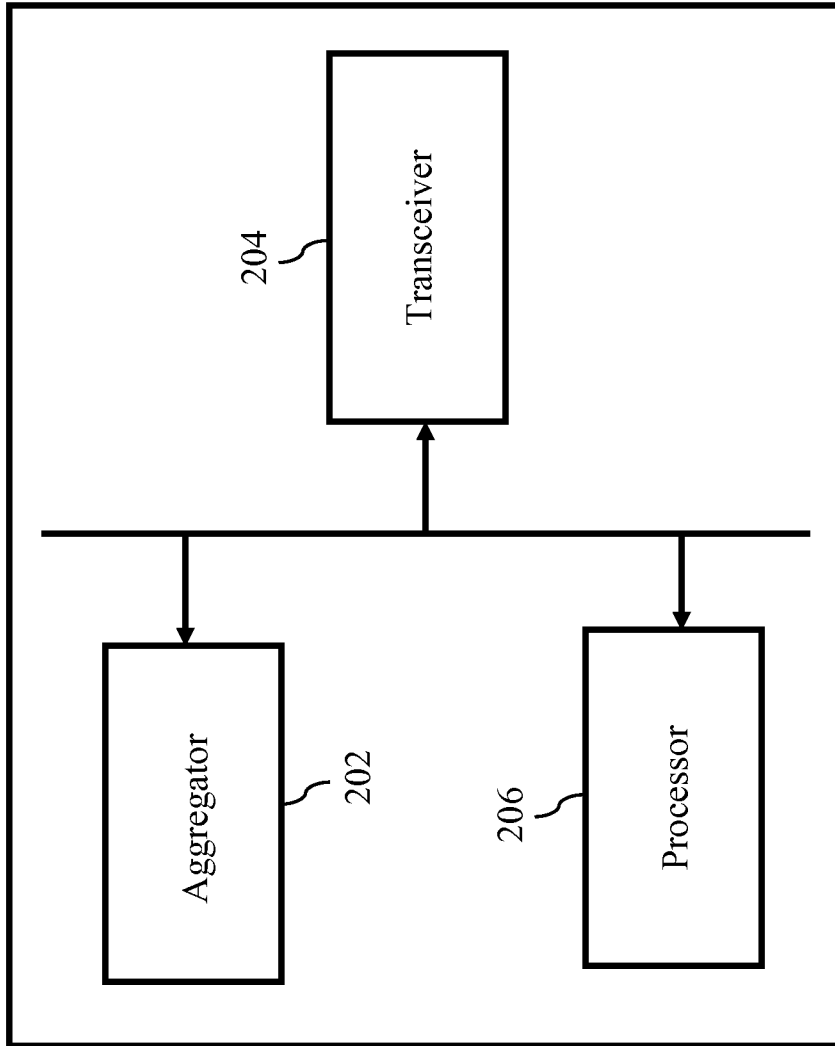


FIG. 1

2/6



200
FIG. 2

3/6

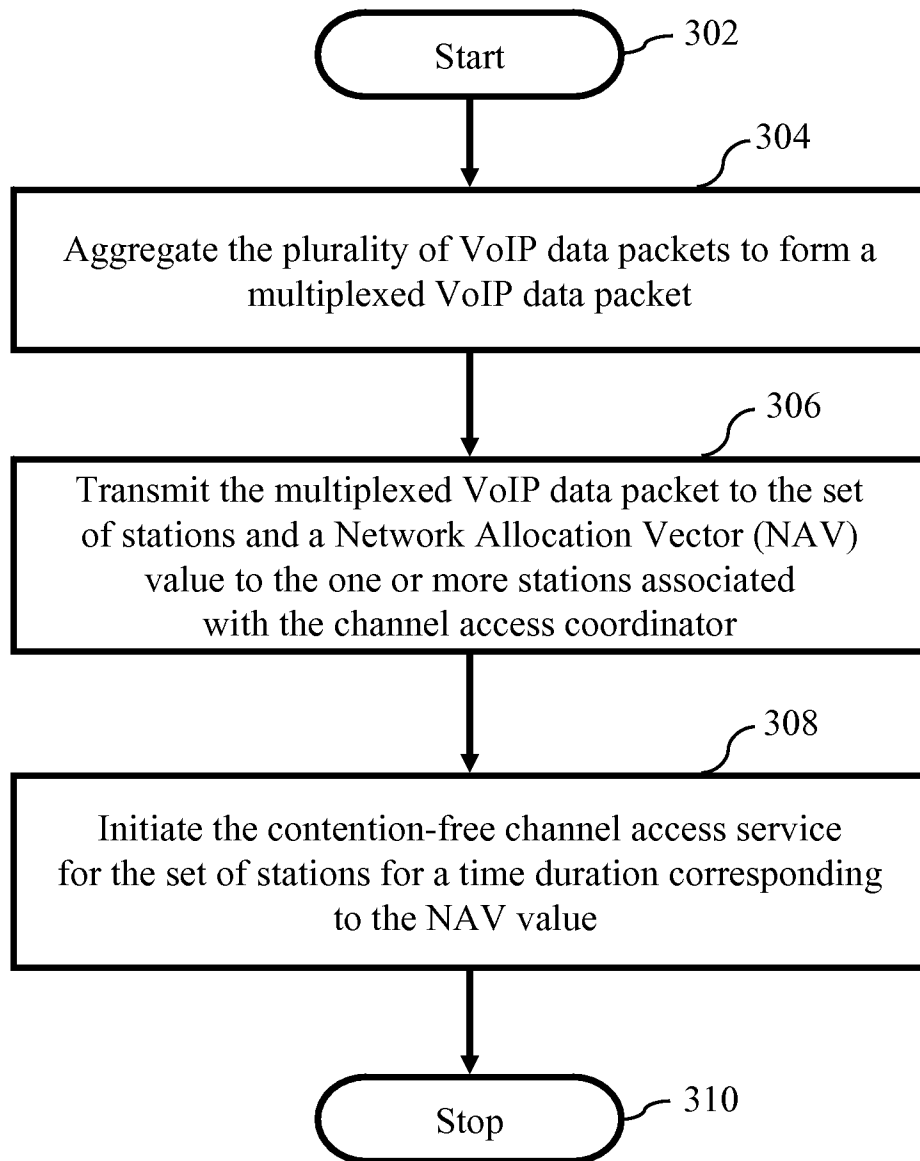
300

FIG. 3

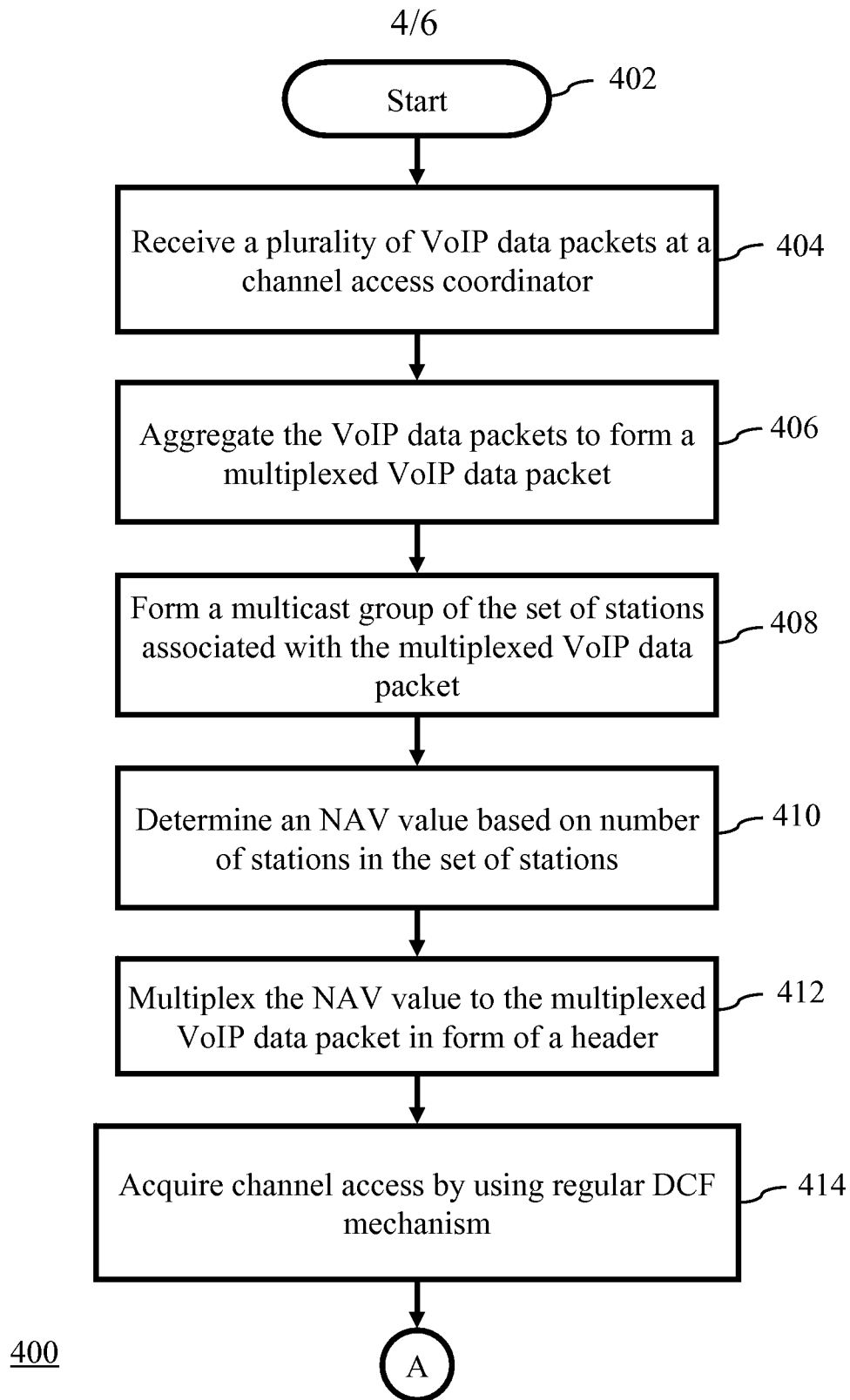


FIG. 4

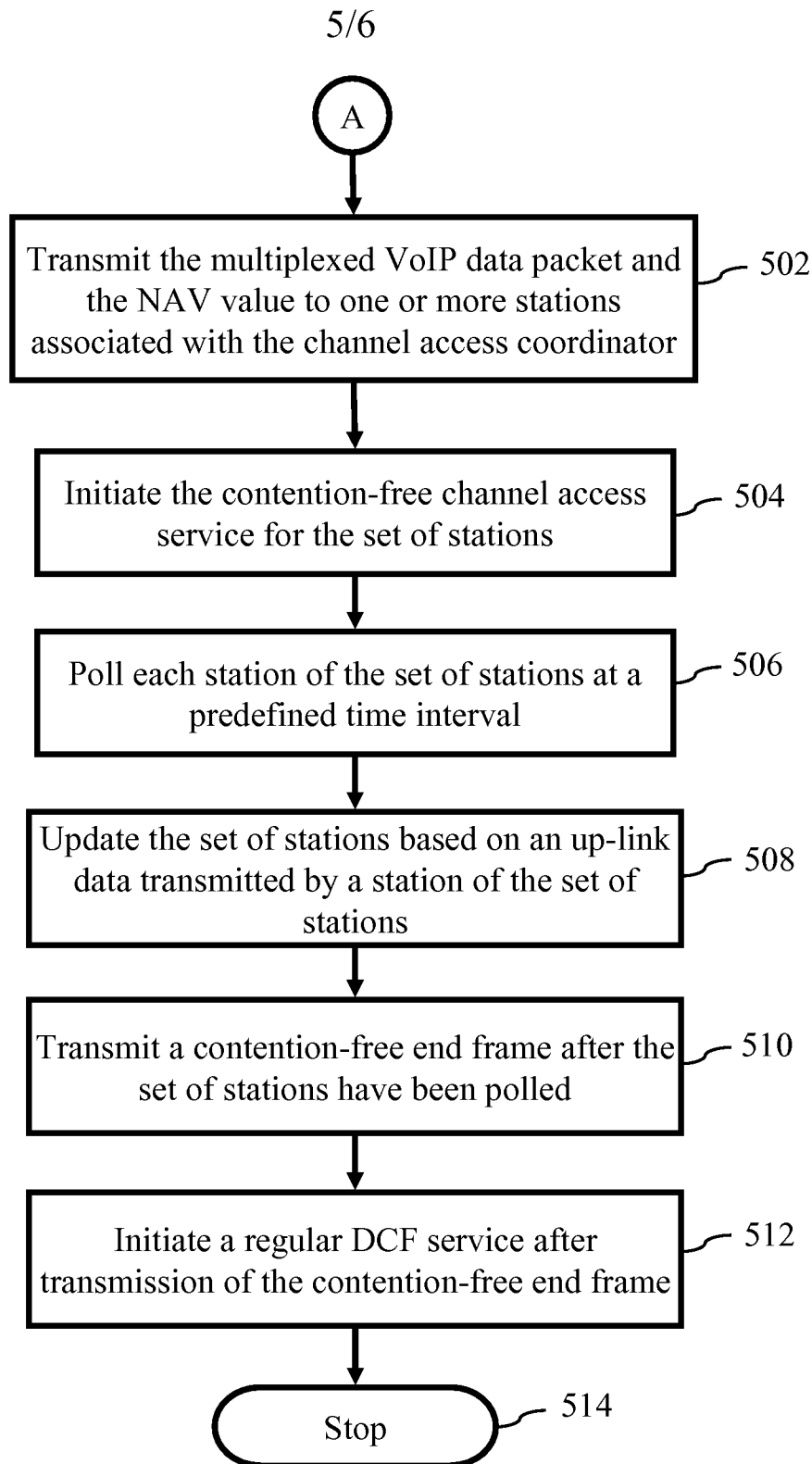


FIG. 5

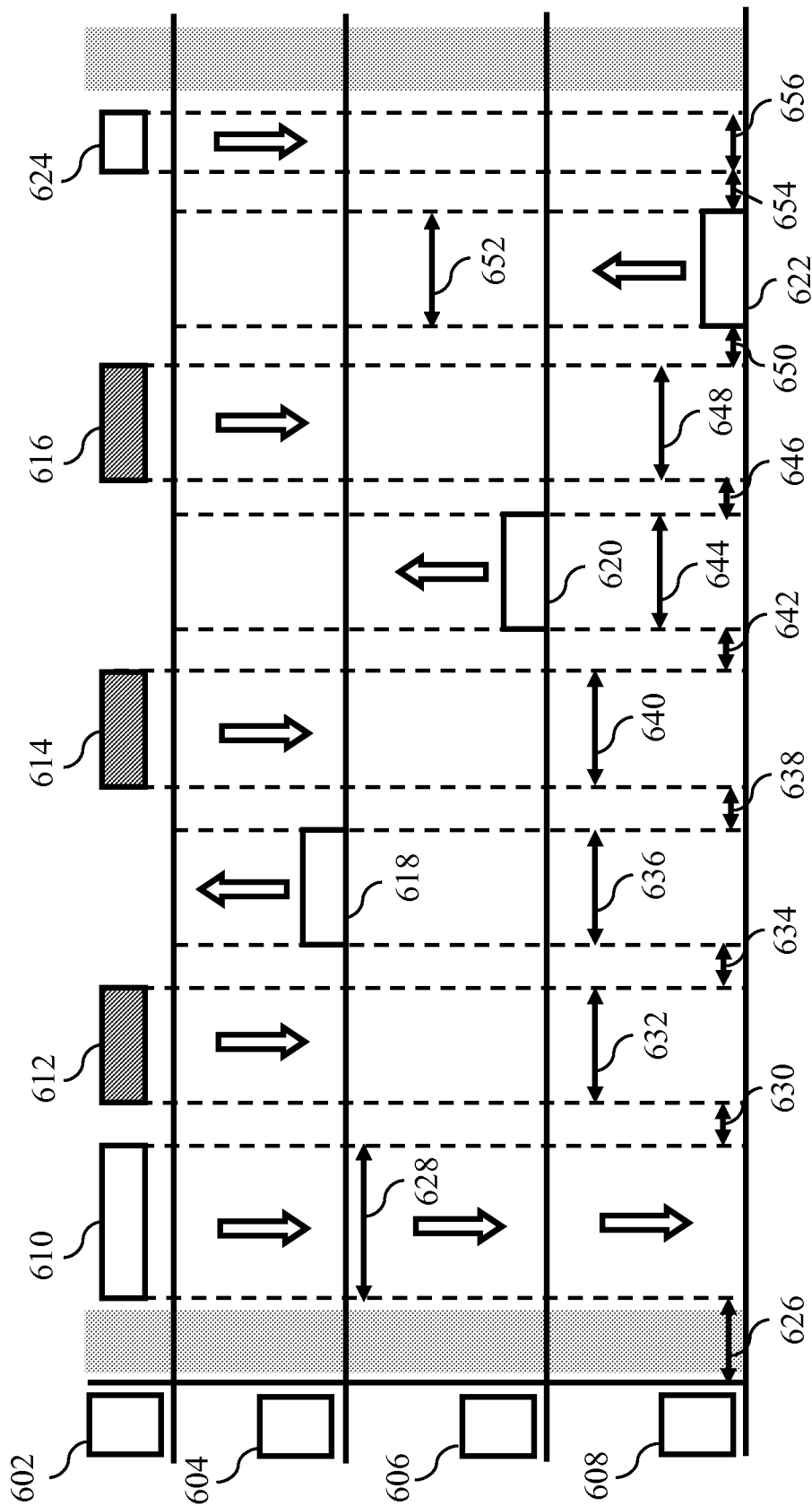


FIG. 6

INTERNATIONAL SEARCH REPORT

International application No.

PCT/US08/61210

A. CLASSIFICATION OF SUBJECT MATTER

IPC(8) - H04L 5/14 (2008.04)

USPC - 370/294

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC(8) - H04L 5/14, 12/66; H04J3/22; H04B7/00 (2008.04)

USPC - 370/294, 469, 352; 455/41.2

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

PatBase; MicroPatent

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 2005/0025182 A1 (NAZARI) 03 February 2005 (03.02.2005) entire document	1, 3, 6, 14-16 and 20
Y		2, 4-5, 7-13 and 17-19
Y	US 2005/0135284 A1 (NANDA et al) 23 June 2005 (23.06.2005) entire document	2, 5, 7-10, 12-13, 17 and 19
Y	US 2005/0124294 A1 (WENTINK) 09 June 2005 (09.06.2005) entire document	4, 11 and 18-19
A	US 2003/0081592 A1 (KRISHNARAJAH et al) 01 May 2003 (01.05.2003) entire document	1-20
A	US 2007/0002749 A1 (SONDUR et al) 04 January 2007 (04.01.2007) entire document	1-20

☐ Further documents are listed in the continuation of Box C.

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search

11 June 2008

Date of mailing of the international search report

27 JUN 2008

Name and mailing address of the ISA/US

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