INFORMATION ELEMENT/GROUP NAME	NEED
Message Type	MP
RB INFORMATION ELEMENTS	
RB Identit	MP

[0056] Once a packet-switched call has been established and is ongoing, the possibility exists for either of the two parties to conference in additional parties. The multi-party call setup is done using SIP signaling (or other call control signaling). Once additional parties have been conferenced in, the downlink RTP packets from the multi-parties typically go through an RTP mixer, where the voice data from multiple sources are mixed into a single RTP packet. In a two-party call, the downlink RTP packet header includes the standard 12 bytes according to RFC 1889.

[0057] However, once additional parties have been conferenced in, the downlink RTP packet header increases in size since additional RTP fields are included in the RTP header. One field that is included is the CSRC count field (CC). If the value of CC is zero, then the call is a two-party call. If the value of CC is one, then the call is a three-party call. Given CC=N, there will be N CSRC identifiers in the RTP packet header. The list of CSRC identifiers is present only when inserted by an RTP mixer. The presence of this list is indicated by the parameter CC. Thus, the radio network controller can continuously snoop or monitor the parameter CC in messages received from the SGSN **24** to determine if the RTP/UDP/IP information in the mobile station needs to be reconfigured due to the presence of additional parties in the call.

[0058] The reconfiguration is performed by use of a DOWNLINK PROTOCOL HEADER RECONFIGURATION message, whose content is provided below:

INFORMATION ELEMENT/GROUP NAME	NEED
Message Type	MP
RB INFORMATION ELEMENTS	
RB Identit	MP
RTP/UDP/IP HEADER INFORMATION ELEMENTS	
DiffServ Code Point (DSCP)	OP
RTP CSRC Count (CC)	OP
RTP Synchronization Source Identifier (SSRC)	MP
RTP Contributing Source Identifier	OP
RTP Sequence Number	MP
RTP Timestamp	MP

[0059] To acknowledge the DOWNLINK PROTOCOL HEADER RECONFIGURATION message, the mobile station sends a DOWNLINK PROTOCOL HEADER RECONFIGURATION COMPLETE message. The content of this message is provided below:

INFORMATION ELEMENT/GROUP NAME	NEED
Message Type	MP
RB INFORMATION ELEMENTS	
RB Identit	MP

[0060] Referring to **FIG. 3, a** call setup procedure according to one embodiment is illustrated. In this embodiment, packet-switched call setup is performed using SIP messaging. The entities involved in the call setup include a mobile station (labeled UE), a radio network controller, the SGSN **24**, the GGSN **26**, the CSCF **40**, the MGCF **42**, and the media gateway **34**. It is assumed that the mobile station UE is the initiator of the call, with the target being a terminal coupled to the PSTN **36**, such as telephone **38** in **FIG. 1**. For purposes of the packet-switched call, the terminating endpoint is the T-SGW **43** (**FIG. 1**) for control signaling and the media gateway **34** for bearer traffic. Packet control signaling and bearer traffic is converted to traditional circuit-switched control signaling and traffic for communication over the PSTN **36**.

[0061] The mobile station first performs a radio resource control (RRC) connection setup (at **100**) with the radio network controller. Next, the mobile station performs a GPRS attach procedure (at **102**). The GPRS attach procedure is performed to inform the radio access network that the mobile station is available. Next, to activate a primary PDP (Packet Data Protocol) context, the mobile station sends (at **104**) an Activate PDP Context request, which is processed by the SGSN **24** and the GGSN **26**. The primary PDP context includes, among other things, the default QoS profile for the requested connection. As part of the primary PDP Context activation procedure, the SGSN **24** performs a radio access bearer assignment procedure to assign one or more radio access bearers to the mobile station.

[0062] After the primary PDP context has been activated, a SIP registration procedure is performed (at **106**). The SIP registration procedure is performed with the CSCF **40**, which includes the SIP proxy. SIP registration is performed to set up the profile for the mobile station in the CSCF **40**, so that the CSCF **40** is aware of the mobile station's existence as well as various configuration information associated with the mobile station.

[0063] After SIP registration, the mobile station can initiate a packet-switched call by sending call setup messages (at **108**). To initiate a call, the SIP INVITE request is sent, which includes the destination address of the terminal being called and indicates that the called terminal is being invited to participate in a call session. Various acknowledgment messages, as defined by SIP, are also exchanged between the mobile station and the CSCF **40**. The SIP messages are routed through the CSCF **40** since the CSCF **40** acts as the SIP proxy.

[0064] Next, the mobile station initiates an activates secondary PDP context procedure (at **110**). A different QoS profile can be assigned in the secondary PDP context to enable a higher level of service if desired for the bearer traffic (e.g., packet-switched speech data). After the activate secondary PDP context procedure, further SIP call setup