

art that the present invention may be practiced without these details and that numerous variations or modifications from the described embodiments may be possible.

[0017] Referring to FIG. 1, a communications network 10 includes a wireless core network 11 that enables communications with mobile stations (e.g., 16, 18, 20, and 22). The wireless core network 11 includes radio access network (RAN) equipment 12 and 14 for communicating with the mobile stations 16, 18, 20 and 22 over wireless links. A wireless link is also referred to as an air interface. The radio access network equipment 12 includes a GSM/EDGE (Global System for Mobile/Enhanced Data Rate for Global Evolution) radio access network (GERAN) system. GERAN provides for enhanced data rates for best-effort services (e.g., web browsing, electronic mail, and so forth) and real-time traffic (e.g., voice-over-Internet Protocol or voice-over-IP). A version of IP, referred to as IPv4, is described in Request for Comments (RFC) 791, entitled "Internet Protocol," dated September 1981. Another version of IP is IPv6, which is described in RFC 2460, "Internet Protocol, Version 6 (IPv6) Specification," dated December 1998.

[0018] The radio access network equipment 14 includes a UMTS (Universal Mobile Telecommunication System) terrestrial radio access network (UTRAN) system. The UTRAN system 14 is based on the wideband code-division multiple access (W-CDMA) technology.

[0019] The GERAN system 12 includes a GERAN base station transceiver (or radio) and a GERAN radio network controller (RNC), and the UTRAN system 14 includes a UTRAN base station transceiver and a UTRAN radio network controller (RNC). More generally, a "wireless access system" refers to any system (such as the GERAN or UTRAN base station transceiver and RNC), implemented as one or plural modules, that is capable of communicating with mobile stations over defined channels on wireless links.

[0020] The GERAN radio network controller is coupled to a serving GPRS (General Packet Radio Service) support node (SGSN) 24 over a Gb link or an Iu link (specifically an Iu-ps link for packet-switched data). Signaling and user data can be communicated between the GERAN radio network controller and SGSN 24 over each of the Gb and Iu links. The UTRAN radio network controller is coupled to the SGSN 24 over an Iu link (specifically an Iu-ps link for packet-switched data). The SGSN 24 (along with the GGSN 26 and the RNC portions of the GERAN system 12 and UTRAN system 14) controls the establishment, processing, and termination of packet-switched communications sessions between mobile stations 16, 18, 20 and 22 and another endpoint.

[0021] The SGSN 24 is in turn coupled to a gateway GPRS support node (GGSN) 26 over a Gn interface. The GGSN 26 acts as a gateway between the wireless core network 11 and a packet network 28, such as the Internet or other type of packet network or even another wireless core network. The GGSN 26 is coupled to an edge or border gateway router (not shown) in the packet data network 28 over a Gi interface. The packet network 28 is coupled to various endpoints, such as a PC telephone 30 and a user station 32 (e.g., a computer system).

[0022] The GGSN 26 is also coupled to a media gateway (MGW) 34 over a Gi interface. The media gateway 34 acts

as a gateway for communications of bearer traffic between (1) the wireless core network 11 and a circuit-switched network such as a public switched telephone network (PSTN) 36 and (2) the wireless core network 11 and the Internet 28 (in the event that transcoding is required for wireless Internet technology to wireless/landline Internet telephone calls). The PSTN 36 is coupled to various terminals 38, such as telephones, and the Internet 28 is coupled to various terminals 30, 32, such as PC telephones.

[0023] The wireless core network 11 also includes a call state control function (CSCF) module 40 that provides call control for a packet-switched communications session. In some embodiments, the CSCF module 40 is a (Session Initiation Protocol) SIP proxy or server that receives call requests on behalf of other entities, resolves logical addresses or identifiers in the call requests, and forwards the call requests to intended destinations. SIP defines a call establishment protocol that can be used to initiate call sessions as well as to invite members to a session that may have been advertised by some other mechanism, such as electronic mail, news groups, web pages, and other mechanisms. A version of SIP is described in RFC 2543, entitled "SIP: Session Initiation Protocol," dated August 1999. In other embodiments, other types of call control protocols or standards can be used, such as the H.323 standard.

[0024] Another module in the wireless core network 11 is a media gateway control function (MGCF) module 42 that provides (1) signaling conversion (e.g., SIP-to-SS7 and vice versa via the MGCF 42 and T-SGW 43 interface) and (2) control of transcoding (e.g., speech data in RTP payload formats-to-PCM transcoding and vice versa in the MGW 34).

[0025] The wireless core network 11 is capable of providing conventional packet data services, such as electronic mail, web browsing, file transfer, and so forth, for the mobile stations 16, 18, 20 and 22. Such data services may be provided for communications sessions between a mobile station and an endpoint coupled to the packet data network 28 or PSTN 36. The wireless core network 11 is also capable of providing packet-switched voice and other real-time communications between the mobile stations 16, 18, 20 and 22 and endpoints coupled to the packet data network 28 or PSTN 36. As used here, "real-time communications" refers to communications in which data is exchanged on a substantially real-time basis between two endpoints (that is, the communication is delay intolerant). Examples of real-time data include voice data exchanged in a call (or telephony) session, video data exchanged in a video conferencing session, and so forth.

[0026] In packet-switched communications, user data such as voice or other types of data are carried in packets, such as IP packets. In one embodiment, real-time data such as voice is converted to a Real-Time Protocol (RTP) format and carried as an RTP payload in a UDP packet that is encapsulated in an IP packet. RTP is described in RFC 1889, entitled "RTP: A Transport Protocol for Real-Time Applications," dated January 1996. RTP defines end-to-end transport functions that are suitable for real-time data, such as audio, video, or other data.

[0027] IP provides network layer functionality (node-to-node routing functionality) for packet-switched communications over a network. Unlike circuit-switched networks,