SIP or H.323: Which Call Control Protocol is More Suitable for the 3G UMTS All-IP Network?

Krisztián Kiss

The today's telephony service is provided mostly over circuit-switched networks, which are referred to as Public Switched Telephone Network (PSTN). In recent years a new trend is beginning to emerge: the telephony service over IP networks, which results the integration of voice and data applications. However, for operation of an IP telephony architecture it is needed to provide a signaling infrastructure, which offers at least the same capabilities and features as the Signaling System (SS7) architecture in PSTN.

The purpose of a multimedia signaling protocol is to enable two-way communication between two or more endpoints. Currently, there are two major candidates for packet-based communication being specified, the H.323 protocol suite from International Telecommunication Union (ITU-T) and the Session Initiation Protocol (SIP) from the Internet Engineering Task Force (IETF). Both protocols define methods how the endpoint can register to a multimedia network, initiate call setup procedure, exchange capability information, and establish, modify and release calls. Both protocols also support multiparty conferencing and supplementary services. Neither SIP nor H.323 has been originally designed for a mobile IP network, and thus architecture supporting the mobility of the endpoints imposes some new requirements on the protocols. The proposal from 3GPP for the Release 2000 UMTS standard [1] has been developed with the goal of allowing operators to deploy an All-IP based architecture to deliver 3rd Generation wireless services. The architecture, which contains the following key segments: radio network, GPRS (General Packet Radio Service) network, call control, gateways to external network, and service architecture, has been kept generic and is not based on a specific call control mechanism such as H.323 or SIP.

H.323 [2] is a standard that specifies the components, protocols and procedures that provide multimedia communication services: real-time audio, video, and data communications over packet networks including IP based networks. The H.323 standard specifies four kinds of components: terminal, gateway, gatekeeper and Multipoint Control Unit (MCU). H.323 system control is provided by three separate signaling functions: the H.245 Control Channel [3], the Q.931 Call Signaling Channel [4], and the RAS (Registration, Admission and Status) Channel [4]. The RAS signaling function uses H.225.0 [4] messages to perform registration, admissions, bandwidths changes, status, and disengage procedures between endpoints and Gatekeepers. The call signaling function uses H.225.0 call signaling to establish a connection between two H.323 endpoints. The H.245 Control Function uses the H.245 Control Channel to carry end-to-end control messages governing operation of the H.323 entity, including capabilities exchange, opening and closing of logical channels, mode preference requests, flow control messages, and general commands and indications.

SIP [5] is an application-layer signaling protocol, which is used to establish and control multimedia sessions or calls, both unicast and multicast. SIP specifies basically two kinds of components: terminals (user agents) and SIP network elements, which can be proxy servers, redirect servers, location servers or registrars. In SIP the gateways are considered as special cases of user agents. The SIP protocol includes basic call signaling, user location, registration and as an extension also advanced signaling. The other services, such as quality of service, directory access, service discovery, session content description and conference control, are orthogonal and reside in separate protocols. SIP has a modular architecture, where different functions are performed in different protocols. Protocols can easily be replaced, and even components of H.323 can be integrated into the SIP environment. SIP uses the Session Description Protocol (SDP) [6] to describe the capabilities and media types supported by the terminals. SIP is independent of the environment, and does not require any reliable transport protocol. In fact any datagram or stream protocol that delivers a whole SIP request or response in full can be used. Such protocols are UDP and TCP in the Internet.

H.323 is an umbrella standard actually covering many protocols, all of which are needed for H.323 multimedia services. H.323 uses binary coding and ASN.1, making stack implementation a big effort. SIP, on the other hand, being lightweight and text-based, is easier to implement and is supported mainly by the Internet community. The complexity of the implementation may be an issue in the terminal, but not so much in the network. In the network, the most important issues are reliability, support for future extensions, and performance. SIP has been said to be more scalable because the SIP call processing functionality implementation can be stateless, thus requiring less memory in the network element. However, new requirements set by the mobile IP telephony network, like charging, statistics, and IN triggering would either require dedicated servers for these purposes, or a stateful implementation, thus resulting in a similar memory consumption than with H.323.

At the moment, neither of the protocols supports the requirements imposed by mobile IP telephony. The basic functionality of the two protocols is quite similar, and with the modifications both could satisfy the requirements of a mobile IP telephony network. Also, should both protocols be selected as alternatives for UMTS Release 00, the interworking of the protocols is not foreseen to be a major obstacle.

References

[1] 3rd Generation Partnership Project, Technical Specification Group Services and System Aspects, Architecture for an All-IP network, 3G TR 23.922 version 1.0.0, October 1999

[2] ITU-T Rec. H.323: Packet-based multimedia communications systems, Ver. 02/98

[3] ITU-T Rec. H.245: Control Protocol for multimedia communication, Ver. 09/98

[4] ITU-T Rec. H.225.0: Call signaling protocols and media stream packetization for packet-based multimedia communication systems, Ver. 02/98

[5] Handley, M., Schulzrinne, H., Schooler, E., Rosenberg, J.: SIP: Session Initiation Protocol, IETF RFC 2543, March 1999

[6] Handley, M., Jacobson, V.: SDP: Session Description Protocol, IETF RFC 2327, April 1998