Comparison of H.323 and SIP Protocol Specification

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Abstract

The two major standards in the multimedia services over IP area are the protocol suites H.323(ITU-T) [1], [2] and SIP (IETF) [3]. Although both standards are quite similar on the basic levels of the protocols architecture, the paper is shown that there are considerable differences on the higher levels regarding their supplementary sevices. SIP is designed with a broader scope, offering functions specifically designed to enable easy extensions, it should be the advantage for new potential services. H.323 is still the more mature standard, H.323 provides better interoperability and interworking (PSTN, ISDN). We can assume a coexistence of both protocols. This paper presents a brief history of the Voice over IP, introduces a comparison SIP and H.323 standard from a technical view.

1. Introduction

H.323 is a umbrela specification, meaning that it is not a protocol by itself, but rather defines how to use other protocols. H.323 was developed by the International Telecommunications Union (ITU) in 1996. Standard consists of several protocols, including above all H.225 RAS signaling, H.225.0 Call signaling (Q.931), H.245 Control signaling, RTP (Real Time Protocol), RTCP (Real Time Control Protocol), H.450 Supplemetary services and other standards for voice and video digitization and compression. H.323 was designed as a standard for realtime videoconferencing on a local area network. [1], [2]

Session Initiation Protocol (SIP) was created by the Internet Engineering Task Force (IETF) as the standard RFC 2543 in 1999. The IETF and the ITU have a very different approach to the development of a protocol specification. SIP is a much simpler protocol that can be used for call set up and management of VoIP applications. SIP provides advanced signaling and control functionality for a large range of multimedia communications. The multimedia session is described in two levels, the description of the media streams that are exchanged between the parties of a multimedia session are defined by protocol SDP. Session Description Protocol (SDP, RFC 2327) in fact is not a protocol, but a structured text-based description format that can be carried in the SIP message body. The using of telephony services can be provide by PINT protocol (PSTN/Internet Interworking, RFC 2848) too. [3], [9]

2. Comparing the Basic Architecture

The two VoIP architectures SIP and H.323 are based on similar concepts. The basic call procedures and feature controls are performed mainly in the terminals. For network support of control mechanisms, servers are required in the VoIP networks, as Gatekeeper (GK) in H.323 resp. Proxy Sever in SIP.

H. 323 elements:
- Terminals
- Gatekeepers
- Gateways
- Multiconference Unit (MCU)

SIP elements:
- User Agents
- Proxy Servers

Realtime Data Transmission:
- H.323: RTP/RTCP
- SIP : RTP/RTCP

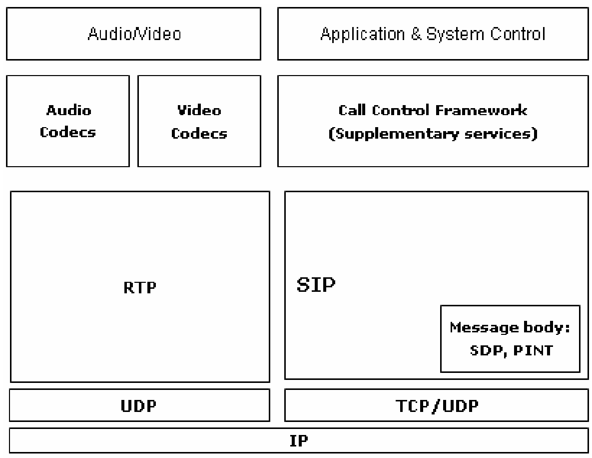

Figure 1.: SIP Protocol Suite.

Call Control:
- H.323: H.225.0/H.245
- SIP : SIP/SDP/PINT

Signaling Procedure:
- H.323: Basic Call Setup or Fast Connect (min. H.323 version 2)
- SIP : SIP-INVITE Transaction

Feature Control:
- H.323 : H.450.x
- SIP : Call Control FrameworkH.323: RTP/RTCP

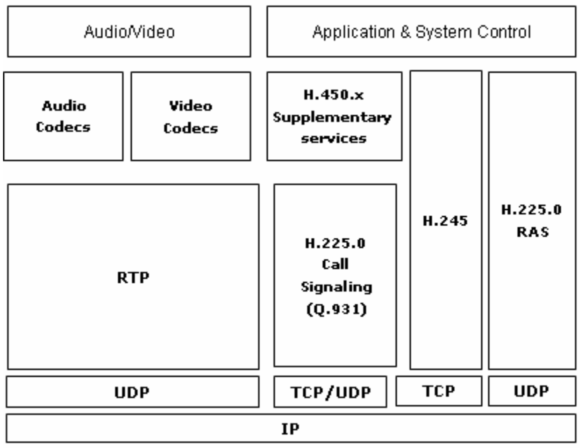

Figure 2.: H.323 Protocol Suite.

The focus of this paper is on comparing the service architecture of H.323 and SIP. The following chapter provide a look into Call Setup Procedures.

3. Comparing the Basic Call Setup

H.323 is a very complex protocol, one of the biggest criticism of H.323 is that it can result in the transmission of many unnecessary messages across the network [4]. The ITU approach generally is to try to anticipate everything that anyone would ever want to do and include as much as possible in the specification.

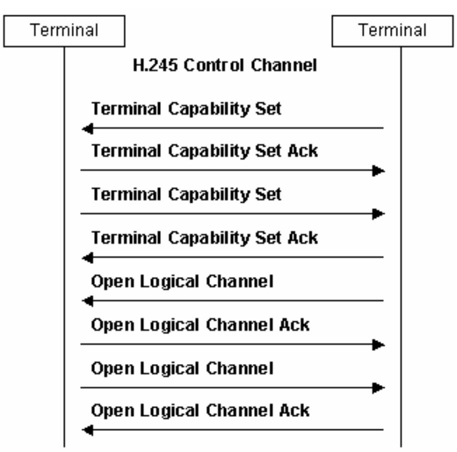

Figure 3. : H.245 Control Channel.

It is reason why H.323 is very large protocol. SIP is working with minimal numbers of messages and as RFC is published much quicker than ITU documents.

H.323 control procedures of connection have three parts, H.225 RAS, H.225 Call Signaling and H.245. H.245 provides control to the multimedia session that has been established and messages are carried via a special channel called the H.245 control channel, see Figure 3.

Openning the H.245 Control Channel is optional, this mode is called as Fast Connect. With the use of Fast Connect, there is no need to open an H.245 channel, the endpoints may transmit fastStart element in the SETUP message and return a fastStart element in any message to the caller.

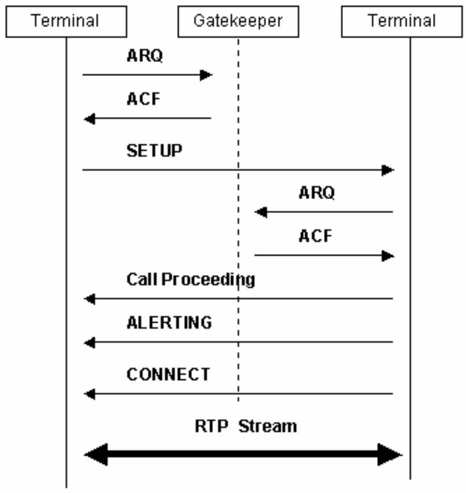

Figure 4.: Basic Call Setup using H.323 Fast Connect

Figure 4 illustrates the message exchange that is required to set up a similar call using H.323 Fast Connect and SIP in Figure 5.

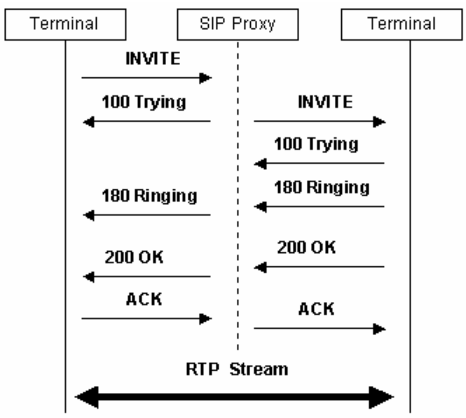

Figure 5.: Basic Call Setup using SIP.

The trace of H.323 connection requires the debugging tools enabling decoding of ASN.1 format, whereas SIP messages use text formats. [7]

4. Conclusion

In my opinion, H.323 is the better choice for VoIP networks, H.323 provides better functionality, interoperability with respect to supplementary services [5], H.323 describes and enables an object-oriented approach based on QSIG in accordance with ITU-T recommendation H.323 Annex M1 11/00 and provides better interworking with already existing telephony systems (e.g. ISDN).

The programming languages derived from HTTP are more easily applicable for SIP, SIP is based on HTTP. SIP is the better for new services implementation. SIP requires less code to implement than H.323.

H.323 was an early leader in the VoIP market, and continues to be used in a lot of networks, but IETF standards are more flexible and accessible then ITU protocols and it could be important for the VoIP device design and its development. The reality is that most H.323 products on the market today also support SIP, including SIP/H.323 interworking.

It can be assumed, that neither of the two protocols will succeed over the other. They will probably coexist in different environments, bringing a strong requirement on interworking between them.

References

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