New Generation Networks Architecture between H.323 and SIP Protocol

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Abstract Today in this new era everyone wants to achieve the goal or target in minimum time as per his schedule, for business purpose, corporate sector, or make a better relationship via meetings either in his personal life. It’s a major challenge for new generation is to provide multimedia teleconferencing services. For this challenge there are special standards have recently populated for signaling and control for Internet teleconferencing. Here discussed on two standards: One is ITU Recommendation H.323, and the other is the IETF Session Initiation Protocol (SIP). Both signaling protocols are responsible for call setup and call tear down. Several comparisons of these two protocols have been published already, but their service architectures have been rarely addressed. This paper provides a unique architecture based on mechanism, with comparison of H.323 and SIP both protocol, and focusing on their service architectures. While architecture of both standards are quite similar. Here in this paper focused on considerable differences regarding their transferable and supplementary service. H.323 is still the more standard, smooth interworking with the PSTN and interoperability between different implementations. It has specific advantages for IP telephony applications. SIP has been designed with a broad scope, providing more generic syntax and semantics regarding feature definition and session description. A coexistence of both protocols can be foreseen, stressing the importance of interworking between them. This paper describes to all differences, properties and provides a unique architecture.

Keywords SIP, H.323, Multipoint Control Unit, CMA Gatekeeper, and Testing of Services

1. Introduction

H.323 and SIP standard provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. Both are apply specially for video conferencing services at broad level in today network. Since for provide useful services in Internet telephony, requires a set of control protocols for connection establishment, capabilities exchange, and conference control for voice and video. Currently, these two protocols exist to meet this need. One is ITU-T H.323, and the
other is the IETF Session Initiation Protocol (SIP). The H.323. By abidance to H.323, multimedia products and applications from multiple sources can interoperate, allowing users to communicate without concern for compatibility. H.323 will be the keystone for LAN-based products for consumer, business, entertainment, and professional applications. In this paper, H.323 and SIP are compared according to the following criteria: standardization status, supported services, supplementary service architecture, and mechanisms, interoperability of services and features, and service creation issues. Basic call features like call setup and session modification are distinguished from supplementary services. First we discuss about both protocols briefly then compared on all activities.

1.1. H.323 Standard

The ITU-T H.323 standard fulfill to all the communicational needs for multimedia system by using packet based network [11]. The network can may be included any types like as Local Area Networks, Enterprise Area Networks, Metropolitan Area Networks, Intra-Networks, and Inter-Networks (including the Internet). There are some related coefficient of H.323 discussed here those mainly associated points. H.323 Terminals: multi endpoint connection those able for real time voice or video communications with other H.323 terminals, gateways or MCUs on the network.

MCU/MC/MPs: for this standard need a central hub which able for manage multi connection, i.e. called Multipoint Controller Units (MCU), which include a Multipoint Controller (MC) and one or several Multipoint Processors (MPs), these all devices worked centrally and control to multi connection and process.

Gateways: Gateway use for provide interconnection between IP networks and Switched Circuit Networks (SCNs), such as ISDN and PSTN. It's used when the two endpoints not connected with same MCU.

Gatekeepers: Gatekeepers play very important role for VoIP services to the endpoints. Mandatory functionality includes address resolution (aliases to IP address mapping), authentication and service authorization. In addition, gatekeepers may offer an array of services such as CDR generation (service accounting for billing), supplementary services (such as call forward, diversion and park and pick-up) and dialing plans.

1.2. SIP Standard

The IETF Session Initiation Protocol (SIP) standard maintain the complete session of video and voice. This is a signaling protocol which initiates, manage and terminate the session across packet networks. The main benefits of use the SIP standard is able to one or more participants at same session or different session. Its SIP supports unicast and multicast communication, borrowing from Internet protocols, such as HTTP, SIP is text-encoded and highly extensible. SIP may be extended to accommodate features and services such as call control services, presence, instant messages, mobility and interoperability with existing telephony systems. Following are the four types of logical SIP entities:

User Agent: User Agent (UA) is the endpoint entity. User Agents initiate and terminate sessions by exchanging requests and responses. The User Agent as an application, which contains both a User Agent client and User Agent server. Devices that could have a UA function in a SIP network are workstations, IP-phones, telephony gateways, call agents, automated answering services and many more.
Proxy Server: It's an interrelationship entity which acts as a server and also as a client for the purpose of making requests on behalf of other clients. Proxy server also works for hold to end points till the response will not be available from other side.

Redirect Server: Its accept SIP request, maps the SIP address of the called or more new addresses and return them to the client after mapped. Unlike Proxy servers, Redirect Servers do not pass the request on to other servers.

Registrar: Its mainly use for register and record to all the operation. SIP has one protocol format for all actions, such as Registration, Call Control, and Presence.

1.3. Basic H.323 - SIP Call Scenario

2. Methodology

2.1. The Basic Protocol Architectures

2.1.1. **H.323 Protocol Architecture** H.323 protocol architecture first time passed by Study Group 16 in December as a first version of this standard H.323 v.1. Those completely capable for established a videoconferencing call on a LAN network [1, 2]. In which used to all the recommended connection and entities like as Gatekeepers, Gateway, and MCU, which provide multimedia communication over packet based networks. This standard is able for both audio and video. After then passed a new version the ITU-T H.320 protocol suite (H.245, H.225.0-CC), which able for used the existing protocol directly RTP and RTCP. This standard was designed a new concept for established the connection RAS (Registration, Admission and Status). RAS signaling functions are required for endpoint registration, admission control and address resolution. Call signaling function includes connection setup, capability exchange and open logical channel procedures, which also useful for maintain the records. H.323 established the end to end point connection by connected with MCU which perform the conference. Here following we define the protocol suit based basic architecture diagram of H.323 protocol, in which all the steps and mechanism will very clear. Diagram related to reference no. [8].
2.1.2. **SIP Basic Protocol** SIP Protocol first time proposed by IETF (Multiparty Multimedia Session Control Working Group), which is capable for control telephony features over wide area network. In this standard, the main benefits are applicable for any type of network and able to integrate stores with conference multimedia. SIP capable for providing advanced signaling and control functionality for a large range of communication. The function of this standard is to locate the parties or resources, finds the actual address and network then sends invitation for service session and negotiation of session parameter. SIP provides a small number of text based messages to be exchanged between the SIP peer entities. By sending the message, servers will able for traverse by network. In this way, baseline SIP according to RFC 2543 includes all basic call control functionality in one signaling transaction using the INVITE request message. Conferences in SIP are normally lightweight multicast conferences, to which a user can be invited. Some Extensions for the management of distributed multipoint conferences have been drafted; in addition to the session processing using SIP, the IETF IPTEL WG proposes several possibilities for the programming of services either for administrators or for the users. SIP session signaling from audio/video media processing is shown in protocol suit. This SIP Protocol suit taken from reference no. [9].
2.2. The Service Architectures

2.2.1. H.323 Service Architecture

In this section, discussed service control of H.323, which is monitored by different models. In this, mainly described Distributed feature control (H.450), Stimulus feature control (H.323 Annex L) and Application layer feature control (H.323 Annex K).

A) Distributed Feature Control Using H.450

This features categorized into three different classes, in which first is local features. Local features can be implemented in the endpoints without requiring specific signaling to other network entities. Examples for local features are: repeat a call, call history and call lists, local address book, and speed dialing, privacy functions like do not disturb and mute, etc. [3]. Next features network-based features are implemented in a centralized fashion in the gatekeeper or as a backend service behind the gatekeeper. Examples are authorization, address resolution, call admission, call detail recording, name/number suppression, etc. The third category of features is the set of supplementary services (H.450). These features are important for special signaling between the corresponding entities. Examples are: call forwarding, call transfer, call completion, call hold, etc.

I. Extension of H.450

H.450 provides different types of process those allow easy extension of feature with define the set. H.450 supplementary service information sent on H.450 APDUs (application protocol data units) that may be contained in any H.225.0-CC message. H.450 APDUs are exchanged between supplementary service entities and does not influence the underlying H.225.0 call state. H.450 APDUs can be extended by manufacturer specific information. H.450.1 provides call-related and calling dependent transport of H.450 APDUs. Further, H.450 defines in a generic way how to proceed with H.450 APDUs that are not supported. This enables interoperability between endpoints with differing feature sets and a stepwise deployment of new supplementary services without having to support them in all endpoints at the same time.

II. H.323/H.450 Architecture Endpoints

The concept of H.450 features can be defined; they can be used together with the basic call as building blocks, which allow conceptual formats as a feature by combination of all small set with carefully designed building blocks. While application and further features are built by using local machine.

III. Building Feature Combinations for 3rd Party Applications

The building features of this architecture creating 3rd party call control applications. The functionality provided via the interface may also include functions like monitoring. The H.323/H.450 building blocks may as a remote controlled. The interface is accessible via a common protocol that may be used CSTA (Computer Supported Telephony Applications). An example for a 3rd party application is ACD (Automatic Call Distribution). It can be built up by combining basic call, call transfer to a music/video server, monitoring an ACD agent.

B) Stimulus Feature Control Using H.323

It is used to control the end point features into the network, defined in H.323 Annex L. Endpoints conforming to H.323 still use functional signaling (H.225.0) for controlling the basic call. This yields basic call interoperability with fully functional H.323/H.450 endpoints. There is little intelligence in the endpoints, the feature logic and the semantic procedures are defined in the centralized feature server. This allows an easier deployment of features only the feature server has to be updated, the endpoints do not have to be changed. On the other hand, there are no standardized semantics for the features. The semantics are implementation dependent. Applications that want to use the feature functionality for 1st party call control need a functional interface (API), which cannot easily be provided using the stimulus approach. A further downside of stimulus feature control is the scalability problem due to
C) Application Layer Feature Control Using H.323

H.323 capable for developing to the new services without any interchange or updating the protocol or end points. This mechanism allows for call-related and call-independent service control. Service control sessions can be maintained between endpoints or between endpoints and the network. In this feature the service control session will be connected after exchange the relevant information such as session ID, URL etc. [6]. The HTTP protocol is used in this service control channel to actually offer, select and activate the services. The service logic is described in HTML pages, scripts, etc. All of these can transfer by HTTP protocol. Thus, features can be controlled from any device running a conventional Web browser.

2.2.2. SIP Service Architecture

SIP session based protocol, its control based on a distributed control model. SIP establishing VoIP connections. It is an application layer protocol for creating modifying and terminating the session with one or more end points. The SIP architecture so similar to HTTP, request is generate by client and forward to the server. For SIP the same feature categories can be applied as in H.323. Local features, network based features like authorization and address resolution in an outbound SIP proxy, and supplementary services. But, in defining supplementary services, SIP started with a different approach as compared to H.323 and standard telephony. Whereas traditional supplementary service implementation is standardized, SIP provides several elements to allow the construction of services. SIP defines the mechanism create the message like Invite, which identify by the headers like a receiver either it will transfer to his destination for proper route discriminators. SIP offers several different possibilities for programming of services with dedicated languages. SIP support session description that allow participants to agree on a set of compatible media type. It’s also support user mobility by proxying and redirecting request to the user current activities. The SIP architecture provides services include- User location, Call Setup, User availability, User Capabilities, call handling. SIP provides a well-defined specification for feature negotiation. When a header is not known by a SIP entity it is ignored without affecting the rest of the request. SIP provides the require header that could be used by a SIP client to make sure in advance that a desired behaviour.

2.2.3. H.323 and SIP Service Architectures

The implementation methods between the SIP and the H.323 service architecture in this survey is based on the following criteria: architecture, protocol extensions, message coding, service programming. The key characteristic of the H.323 service architecture is its explicit definition of separate state machines for each feature independent from the basic call state machine. From the signaling point of view the function split of feature control into framework and extensions is a consequence of this separation. In this SIP architecture, we assume two configurations are used for the call scenarios. One is basic configuration, which includes an H.323 and a SIP block. The other configuration contains an H.323 block and a SIP server block, which reside respectively in an H.323 zone for address resolution and admission control, and in a SIP administrative domain for pre-call registration service and address resolution.

As a consequence, important challenges like the subsequent integration of new features into a running system, the interoperability with heterogeneous endpoints. While according to the SIP baseline SIP features are not explicitly signaled and may even be hidden in the carried session description. Several programming languages are defined in the context of SIP for the programming of SIP servers. Although missing in H.323, they could also be applied for it. Since these programming languages are derived from the HTTP context they are more easily applicable for SIP as SIP is based on HTTP. To activate the appropriate service in the route IP network, SIP can specify the proxy servers that must be traversed by a SIP message; in H.323 this is not possible.
3. Conclusion and Future Work

This survey has given overview architecture and the mechanisms to develop services using the two standards. Although both protocols may be used for video and voice applications. Both standards are design with a different specific features and assets. SIP implements for control the session for end points and also for create the session. H.323 has been set to handle voice and multimedia calls including supplementary services. In this paper, we have modelled and verified the system of interworking between H.323 and SIP with different configurations. H.323 describes and enables an object-oriented approach based on separating supplementary services from basic call control.

As per supporting on voice and multimedia over IP including supplementary services, H.323 is the good choice for IP telephony applications. This includes replacement scenarios for legacy, but is especially true when IP telephony supplements and coexists with legacy telephone systems. H.323 becomes more important for carrier implementations. Although the two standards are approaching each other, their focus and applicability is still different. It can be expected that neither of the two
protocols will succeed over the other. They will probably coexist in different environments and implementations over a longer time, putting also a strong requirement on interworking between them. Here in this we analyses and discuss too completely onto the two standards protocol H.323/SIP, which completely beneficial for the today era in voice and video calling. We consider the use of scripting within the gateway logic as promising approach that should be discussed as work in progress.

References


