Facilitating the Interoperability among Different VoIP Protocols with VoIP Web Services

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Abstract

With the development of the internet the VoIP technology is becoming a popular service on the internet platform. Several VoIP protocols have been suggested to realize the VoIP communication. But different VoIP protocol clients cannot interoperate directly. This paper suggests a VoIP Web service infrastructure to facilitate the interoperability among different VoIP protocols. With this infrastructure all VoIP protocols signaling handling details will be hidden through the corresponding web service abstraction. Therefore different VoIP protocols can utilize the common web service interfaces to achieve the interoperability.

Keywords: VoIP, Web Service, Interoperability, H.323, SIP.

1. Introduction

Now internet is not only an information platform but also a convergent services platform. More and more services are utilizing this platform to serve different customers. Voice over IP (VoIP) is a technology to transport voice communication over IP network such as internet. This technology provides the capability of making phone calls over the packet switched networks instead of traditional circuit switched networks. Because of the low cost feature of the internet usage, VoIP can greatly reduce the telephone call costs comparing with the traditional PSTN system. Therefore VoIP is attracting more customers from the traditional telephone communication area, and more and more companies prepare to invest in the development and utilization of the VoIP systems. It is even expected that the VoIP will completely replace the circuit switched PSTN system in the future.

Since the VoIP technology was first developed, many VoIP protocols have been proposed. H.323 can be considered as the first generation VoIP protocol. H.323 [1] is an “umbrella” specification proposed at first in 1996 by ITU-T and has been updated several times. It suggests a standard for multimedia communication over packet switched network such as LANs, WANs and internet. Now it is one of the most accepted VoIP standards utilized by many VoIP product vendors. The Session Initiation Protocol (SIP) [2] is another VoIP standard proposed by IETF. It includes a suit of call setup and media mapping protocols for multimedia communication over the packet switched network. Due to its simplicity, flexibility and its modular character SIP is becoming one of the most promising VoIP protocols which has attracted many VoIP vendors to provide SIP based VoIP solutions.

From the telephony communication history point of view, the PSTN can be considered as the traditional communication method, whereas comparing with SIP, H.323 can be considered as “legacy” VoIP protocol, although H.323 is still widely used. With the VoIP technology development the old standards and systems will be replaced by the new ones sooner or later in the future. But it is very important for the customers to protect their capital that has already been invested in the existing telephony service delivery infrastructure. This means that the traditional PSTN systems and the “legacy” protocols will coexist in the long run with the new systems and protocols. But the problem is how to reach the interoperability among these protocols. Standards and products have been suggested and developed to achieve the interoperability. In order to
realize the interoperation between the VoIP clients and the PSTN pots, a Gateway is used in the H.323 infrastructure. The Gateway can interconnect the IP based network with the traditional telephony network. Another solution for the interoperation between different VoIP protocols is also utilizing a signaling Gateway to translating VoIP signaling in real time. [3] suggests a signaling Gateway to translate SIP signaling from the SIP clients to H.323 signaling and vice versa. The above interoperability solutions can only solve the interworking problem between two different protocols. They are not able to be extended to accommodate new protocols. In this paper we suggested a common VoIP web service infrastructure which can facilitate the interoperability among different protocols. The interoperability between POTS phones and VoIP clients, and the interoperability among different VoIP protocols can all be achieved under this infrastructure. We also propose a VoIP signaling independent VoIP client (called Simple Client) in the infrastructure to provide a multi-protocol client which can communicate with different VoIP protocol clients without integrating any related protocol into the Simple Client [4]. The rest of this paper is organized as follows: section 2 describes some related works, section 3 illustrates the VoIP web service infrastructure, section 4 explains the interworking modes among different protocols, section 5 introduces the prototype system of the VoIP web service infrastructure, and section 6 gives conclusions and future work considerations.

2. Related Works

Interoperability between two different protocols has been discussed in several documents. [3] suggests a Signaling Gateway to translate the signaling between SIP protocol and H.323 protocol. For the interworking between H.323 and PSTN a Gateway, which includes Media Gateway, Media Gateway Controller and Signaling Gateway components, is suggested to handle media flow conversion and control, and signaling translation by the ITU-T. [5] has also discussed about the interworking issues between SIP and PSTN with Gateway. All these above solutions are based on the one to one solution.

3. VoIP Web Service Infrastructure

Web service is an emerging middleware technology based on XML and Web. It provides standard protocols to describe, publish, discover and invoke the distributed components. Because of its platform and language independent character web service can facilitate heterogeneous interoperability. Figure 1 illustrates the VoIP web service infrastructure.

The VoIP web service infrastructure consists of three layers: Simple Clients Layer, VoIP Web Service Manager Layer and VoIP Protocol Web Service Layer. The Simple Clients Layer consists of so called Simple Clients which integrate only media processing and transmission related protocols. Usually the audio codecs such as PCMU, G.723 etc, and the RTP [6] and RTCP protocols are integrated into the Simple Client. The signaling related protocols are separated from the Simple Client to be implemented as signaling web services which are integrated in the VoIP Protocol Web Service Layer. For example, the protocols such as H.225 and H.245, which belong to the H.323 protocol suit and relate to signaling handling, are integrated into the H.323 Web Service. With separating of the VoIP signaling related protocols the Simple Client can function as different VoIP protocol client through calling different VoIP signaling web service. The VoIP Web Service Manager Layer is a multi-function layer in the VoIP web service infrastructure. It provides authentication service to control the usage of the VoIP web service system. The accounting service can collect the VoIP resource usage information for the billing purpose. Other management services such as billing service can also be easily integrated into this infrastructure. The Call Handler is a call session management core service. It creates, monitors, deletes each call session, and it is responsible for the message dispatching, event forwarding etc. It is the bridge among the different VoIP protocol web services.
The VoIP Protocol Web Service Layer consists of different VoIP protocol web services. Each VoIP protocol web service exposes common VoIP abstraction interfaces. The common VoIP abstraction interfaces encapsulate the different signaling processing details. With the common VoIP abstraction interfaces different VoIP signalings can interoperate with each other, and the Simple Clients can also call different type clients with web service methods. New VoIP protocols can be integrated into this infrastructure easily if it can also be encapsulated to provide the same common VoIP abstraction interfaces. According to our analysis of different VoIP protocols following common VoIP abstraction interfaces are defined to provide the VoIP web services.

- **initCall**
  Prepare a call. Include checking the authorization, resource availability etc.
- **makeCall**
  Send a call setup request to remote endpoint.
- **acceptCall**
  Accept a call setup request from remote endpoint.
- **denyCall**
  Deny a call setup request from remote endpoint.
- **endCall**
  Stop the already connected call.
- **getCallInformation**
  Retrieve the call related information such as RTP IP address, port number, codec etc which collected during the signaling handling process.
- **callProceeding**
  Indicate the signaling handling is in progress.
- **callConnect**
  Indicate call setup success, now the voice over RTP can begin.

4. Interworking under the VoIP web service infrastructure

Figure 2 shows the interconnection model with the VoIP web service infrastructure. All the VoIP web services integrate signaling handling related functions. They can be considered as signaling gateways. Other components such as gatekeeper in H.323, SIP proxy in SIP, and Media Gateway (MG) and Media Gateway Controller (MGC) are needed to cooperate with the VoIP web services for interworking among different clients. Since RTP is the protocol for media flow transmission for different VoIP protocols, the Simple Client also integrates the RTP protocol to transmit the media flow.

With the Figure 2 illustrated model different type VoIP clients and POTS phones can interwork under the VoIP web service infrastructure. In this model a VoIP Web service can act as both web service consumer and Web Service provider. A web service playing which kind of role depends on whether it calls other web service or its service is called. For example, if H.323 Web Service represents a H.323 client to call an SIP client, the H.323 Web Service will request the SIP Web Service to setup the call to the SIP client. In this situation the H.323 Web Service acts as a web service consumer whereas the SIP Web Service acts as a web service provider.

Below we explain the interworking mechanism under the above described VoIP web service model.

4.1 Address Translation

In the VoIP web service infrastructure there exist different types of clients. Different address schemes are used to identify the clients. The H.323 address can be presented as IP address, host name, email address, URL, E.164 telephone number etc. The SIP address can expressed as URL or SIP URI which can in the form of telephone number, user name etc. For the POTS phones E.164 telephone number form is used. For the Simple Client a URL scheme is defined to be compatible with different address schemes. It can support H.323, SIP addresses and also E.164 telephone number. The Simple Client can call different clients with URLs such as “h323:”, “sip:”, “tel:”.

Address translation between Simple Client address and other protocol address is simple, since the Simple
Client provides other protocol compatible address scheme. In the VoIP web service infrastructure the address translation is executed by the VoIP Web Service Manager. The address translation between SIP and H.323 can take the mechanism in [3]. For the call between the VoIP client and the POTS phone, the E.164 address scheme should be used by both sides to omit the address translation process [11]. During the address translation process the Gatekeeper, SIP proxy and VoIP Web Service manager are needed to resolve the addressed for different protocols.

4.2 Connection Establishment

Following give some scenario examples to explain how to realize the interworking between two different VoIP clients. These examples do not cover all call setup possible situation, but the similar scenarios can be deduced from these examples.

- **Interworking between Simple Client and VoIP client**

Figure 3 depicts the call setup process when Simple Client calls H.323 client.

![Figure 3. Simple Client calls H.323 client](image)

When a Simple Client wants to call a H.323 client, it sends the makeCall request to the Call Handler which will forward the request to the H.323 Web Service. The makeCall request includes call setup needed information such as the destination of the call, IP address and port number of the Simple Client for RTP data receiving, and media capabilities of the Simple Client. The H.323 Web Service will now represent the Simple Client to handle the H.323 call setup signaling. Steps 2, 3, 4 are the normal H.323 call setup signaling handling process in case the H.323 client accepts the call request from the Simple Client. After the connection established all media transmission information such as RTP IP address, port number and codec of the H.323 client are collected, the H.323 Web Service sends callConnect message to notify the Simple Client to begin media transmission with RTP. The processes of Simple Client calling other clients are similar except the corresponding VoIP web service will be called to handling the signaling. If a VoIP web service receives a call setup request to a Simple Client from other VoIP client, the request will be forwarded to the Simple Client. If the Simple Client accepts the call, the acceptCall response, which includes call setup information as in makeCall, will be sent to the VoIP web service to continue the call setup process. With the support of different VoIP web services the Simple Client can communicate with different VoIP clients. The Simple Client needs not handle the protocol related signaling, this task is handed to the corresponding VoIP web service. The Simple Client needs only to handle media processing and transmission related protocols. With this mechanism the Simple Client can act as a multi-protocol VoIP client.

- **Interworking between SIP client and H.323 client**

Figure 4 depicts the call setup process when a SIP client calls a H.323 client.

When an SIP client requests to call an H.323 client, it sends an SIP INVITE message to the SIP Web Service which will then represent the SIP client to setup the call with the H.323 client. Next the SIP Web Service sends makeCall message to the H.323 Web Service which will then represent the SIP Web Service to call the H.323 client. In succession the normal H.323 call setup signaling handling process begins as illustrated...
in Figure 4 from Step 3 till 5. After the call connection established the H.323 Web Service sends callConnect message to the SIP Web Service which will then send an SIP 200 (OK) message to the SIP client. After the SIP client sends the ACK message to the SIP Web Service all media transmission needed information is available, and the SIP client and the H.323 client can directly transmit media flows with each other using the RTP protocol.

When a H.323 client wants to call an SIP client, the H.323 Web Service and the SIP Web Service will also intermediate the signaling translation. One thing should be remarked is that without Fast Connect mechanism the H.323 client’s capabilities can be collected only after the “Capability Exchange” process, whereas in SIP the capabilities are included in the INVITE message. The [3] suggested method can be used to solve this problem: after the SIP Web Service receives the makeCall request, it sends an INVITE message without session description, and the capabilities of the H.323 client will be sent through the ACK message to the SIP client after the “Capability Exchange” process are carried out and the H.323 client capabilities are collected.

- Interworking between POTS phone and VoIP client

Figure 5 depicts the call setup process when a POTS phone calls an SIP client.

When a POTS phone calls an SIP client, its call request will be sent to the PSTN Web Service through the PSTN. The PSTN Web Service then represents the POTS phone to call the SIP client. It sends makeCall message to the SIP Web Service which in turn sends an SIP INVITE message to the SIP client. Since the POTS phone and the SIP client use different media processing mechanism, a Media Gateway must be introduced to do the conversion between the two media flows generated by POTS phone and the SIP client. Therefore the Media Gateway is the endpoint for both POTS phone and SIP client. The call setup information in the INVITE message, such as IP address and port number, codecs etc, is that of the Media Gateway. If the SIP client accepts the call an SIP 200 (OK) message will be sent to the SIP Web Service. After the SIP Web Service sends an ACK message to the SIP client, then it sends callConnect message to the PSTN Web Service which in turn forwards the connection established notification to the POTS phone. In succession the media flow transmission begins between the Media Gateway and the SIP client with RTP protocol, and the Media Gateway will be responsible for media flow conversion and forwarding for the POTS phone.

Although in H.323 a Gateway model has been defined to realize the interworking between the H.323 client and the POTS phone, and in [5] a SIP PSTN Gateway is used to bridge the SIP clients and POTS phones, the PSTN Web Service provides a VoIP protocol independent mechanism to support different VoIP protocols. Under the VoIP web service infrastructure the VoIP PSTN Gateway model can still be kept without affect the interoperability between the POTS phone and other VoIP protocols. For example, a POTS phone calls an SIP client through an H.323 PSTN Gateway. The idea is the H.323 PSTN Gateway acts as a normal H.323 client to call the H.323 web service, with the H.323 and SIP interoperability provided by the VoIP Web Service infrastructure the H.323 PSTN Gateway can calls the SIP client, therefore the connection between the POTS phone and the SIP client can be built.

5. Prototype implementation

According to the VoIP web service infrastructure we have developed a VoIP web service prototype system. This prototype system realizes the Simple Client, authentication service, Call Handler, H.323 Web Service and SIP Web Service.

The open source project OpenH323 [7] is chosen to implement the H.323 Web Service, whereas NIST SIP [8] is chosen to implement the SIP Web Service. We use JMF [9] to realize the media processing and transmission related functions. JWSDP [10] is chosen to realize the VoIP Web Services. Figure 6 illustrates the VoIP Web Service prototype system architecture.
Figure 6. VoIP Web Service prototype system architecture

In Figure 6 illustrated architecture the Simple Client provides Login function to authenticate the user, and Make Call function to call different client, and Setting function to adjust Simple Client’s configuration. The VoIP Web Service Manager provides Authentication Service, Registration Service and Accounting Service. The CallHandler provides call scheduling of the system. The Local Call Handler processes the call between the Simple Clients which in the same management domain whereas the Remote Call Handler processes the call between the Simple Client in local domain and the other Simple Client in remote domain. The H.323 Call Handler processes the call between Simple Client and H.323 client, and the SIP Call Handler processes the call between Simple Client and SIP client. In the VoIP Web Service container each VoIP protocol related web service uses a signaling server to adapt to the corresponding protocol clients, and the corresponding web service encapsulate the signaling handling detail of the signaling server to provide unified web service interfaces.

6. Conclusion and future works

In this paper we have presented a VoIP Web Service Infrastructure which can be used as a general model to facilitate the interoperability among different VoIP protocols. We have used several different scenarios to explain the interoperability among different VoIP clients. We have also given description about the prototype system. With this infrastructure all VoIP protocols signaling handling detail will be hidden through the corresponding web service abstraction. Therefore different VoIP protocols can utilize the common web service interfaces to achieve the interoperability.

What we have discussed in this paper is a framework for the interworking among different VoIP protocols. In our prototype system implementation only the Simple Client has realized the interoperability with different types of VoIP clients. It will be extended to support the interworking between H.323 client and SIP client, and the interworking between H.323 client and POTS phone etc. Other consideration such as security and additional services should also be considered in the system.

7. References