Abstract — As a IP telephony protocol, the SIP is gaining increasing popularity. There is yet POTS(Plain Old Telephone Service) line as the existing subscriber line. The Megaco/H.248 may be used for call between the POTS line and SIP. So, it is needed that method for interworking between Megaco/H.248 and SIP. In this paper, we propose the implementation of Residential Media Gateway Prototype in Interworking between Megaco/H.248 and SIP. We had designed and implemented MG prototype platforms for interworking between POTS line and IP network in interworking of Megaco/H.248 and SIP protocol. And, we had found that it is feasible in the real network.

Keywords — MG, MGC, Megaco/H.248, SIP

1. Introduction

Most information conveyed over public telecommunication networks is voice. To do this, circuit-switched networks are employed. While circuit switching provides adequate voice quality, it can be highly inefficient. In contrast, the Internet’s packet-switched networks are much more efficient but ill suited for voice without judicious implementation. Recently, because of the popularity of Internet, the increment of network bandwidth, and the improvement of voice compression technologies, the VoIP has an exploding growth and makes a lot of changes to our lives. From the point of view of a consumer, VoIP makes possible advanced services, like video, that are unavailable from the Public Switched Telephone Network(PSTN) at any cost. And, from the point of view of a telephone company, VoIP improves network efficiency - that is, a packet-switched IP network can handle more calls with the same transmission infrastructure that the PSTN can with its circuit-switched TDM approach.

VoIP can be implemented in several ways. A PSTN-based telephone can communicate with a VoIP application, and vice versa. These telephones can also communicate with each other where part of the call is routed over the Internet instead of solely over a dedicated circuit. Finally, two VoIP applications can communicate directly without accessing the PSTN. As a IP telephony protocol, the SIP is gaining increasing popularity. There is yet POTS(Plain Old Telephone Service) line as the existing subscriber line. The Megaco/H.248 may be used for call between the POTS line and SIP. So, it is needed that method for interworking between Megaco/H.248 and SIP.

In this paper, we propose the implementation of Residential Media Gateway Prototype in Interworking between Megaco/H.248 and SIP.

2. The Overview of Megaco/H.248

Figure 1. Decomposed gateways

In a conventional H.323 gateway, the same device provides the media conversion from TDM to packet also provides the signaling interwork. A new class of gateways is being designed that splits the media handling function from the signaling function. These ‘decomposed’ gateways assign the media functions to a ‘media gateway’ and the signal handling functions to a ‘media gateway controller’(Fig 1). A device which handles call control is the Media Gateway Controller(MGC) and a device which handles media and bearer control is the Media Gateway(MG). This separation created the need for a new class of protocol to link the two devices. It seems that The Media Gateway Control Protocol(MGCP)[1] is limited deployment for that purpose.

A new protocol, Megaco/H.248 protocol, has been standardized by the IETF Megaco Working Group working together with ITU-T Study Group 16, which has been published as both a standards-track RFC and ITU-T Recommendations H.248[2][6][7]. This protocol is based on master/slave protocol, extends media gateway control to include a transport-independent connection model, supports for more advanced services such as multimedia conferencing, and supports for operation in countries around the world. The master/slave approach enables centralization of application intelligence in relatively fewer control servers, highly cost-and performance optimized gateway devices, shorter time to market for new feature introduction across diverse networks, and superior legacy support. This approach also provides flexibility in new service deployment since only the
control servers are updated, and services can be more dynamically added or removed as required. Master and slave elements may be distributed within the customer network as appropriate and are not required to be physically co-resident. These features provide overall cost optimization, reduced time to deployment, and reduced operational costs across the system. The Megaco/H.248 protocol is one of the master/slave protocols.

Megaco/H.248 uses a simple, powerful connection/resource model to describe the logical entities or objects within the MG that can be controlled by the MGC. It is fundamentally based on two key concepts: termination and context. Terminations identify media flows or resources, implement signals and generate events, have properties and maintain statistics. They can be permanent (provisioned) or transient (ephemeral). All signals, events, properties and statistics are defined in packages which are associated with the individual terminations. to support multi-media. The command structure of Megaco/H. As shown in Figure 2, context refers to associations between collections of terminations (T), defines the communication between the terminations, and acts as a mixing bridge.

![Figure 2. Megaco/H.248 connection model](image)

<table>
<thead>
<tr>
<th>Command</th>
<th>Requestor</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Add</td>
<td>MGC</td>
<td>Add a termination to a context, giving it property values specified in the command</td>
</tr>
<tr>
<td>Modify</td>
<td>MGC</td>
<td>Change the characteristics of an existing termination, which can be in the null context</td>
</tr>
<tr>
<td>Subtract</td>
<td>MGC</td>
<td>Remove a termination form an existing context</td>
</tr>
<tr>
<td>Move</td>
<td>MGC</td>
<td>Move a termination from its previous(non-null) context to the one associated with the action within which the command resides and modify its properties as specified in the command</td>
</tr>
<tr>
<td>AuditValue</td>
<td>MGC</td>
<td>Determine the characteristics of a termination or the MG as a whole</td>
</tr>
<tr>
<td>AuditCapability</td>
<td>MGC</td>
<td>Determine the possible values supported for the characteristics of a given termination or the MG as a whole</td>
</tr>
<tr>
<td>Notify</td>
<td>MG</td>
<td>Tell the MGC that one or more events for which reporting has been enabled have occurred on the given termination</td>
</tr>
<tr>
<td>ServiceChange</td>
<td>Either</td>
<td>For individual terminations: tell the responder that the service state has changed. Also used to negotiate a new control session or a change in control</td>
</tr>
</tbody>
</table>

![Figure 3. Commands supported by Megaco/H.248](image)

There are only seven commands (Fig. 3) — Add, Subtract, Modify, Move, Notify, AuditValue/AuditCapability, and ServiceChange — and they all operate on terminations in a consistent manner.

In Fig. 4 provided a VoIP network using Megaco/H.248. It includes two Residential MG(R-MG), one MGC. Both the R-MGs attach one analogue phone set.

### 2.1 Call Setup using Megaco/H.248

Fig. 5 shows the steps that Analogue Phone A makes a call to Analogue Phone B.

1. MGC generates the Modify message towards both the R-MG1 and R-MG2 to check for offhook on the terminations.
2. The Analogue Phone A goes off hook.
3. R-MG detects offhook event and sends the Notify message towards the MGC.
4. The MGC issues a Modify command with event descriptor for detecting dial tone and checking digit-map.
5. The MG plays Dial Tone call progress tone on analogue interface.
6. The Analogue Phone A pushes dialing digits.
7. The R-MG1 detects the digits dialed and sends the Notify command with dialing digits parameter.
8. The MGC generates a transaction with two Add commands to create terminations.
9. The R-MG1 creates two terminations and responds to the Add commands.
10. The MGC generates a similar transaction like that of R-MG1 towards the R-MG2. Also, the transaction includes a signal to play ringing tone and an event to check onhook on analogue line.

11. The R-MG2 plays ringing tone towards Analogue Phone B.

12. The R-MG2 creates two terminations and response to the Add commands.

13. The MGC issues a Modify command with signal descriptor for playing ringback tone.

14. The R-MG1 plays ringback tone towards Analogue Phone A.

15. ~ 16. The R-MG2 detects offhook event and sends the Notify command with offhook event.

17. ~ 18. The MGC generates a transaction towards R-MGs with two Modify commands in one action.

Both R-MG1 and R-MG2 send/receive the RTP media flow including voice.

### 3. The Overview of SIP

SIP is a rendezvous protocol for finding users and setting up and modifying multimedia sessions. VoIP is only one of many possible applications for SIP. Since March 1999, when SIP[3] was accepted by the IETF as a proposed standard, SIP has gained a wide following. The strengths of SIP lie in its simplicity and basic assumptions:

**Component reuse**

In many ways, SIP can be considered a child of the Simple Mail Transfer Protocol (SMTP, RFC2821) and HyperText Transfer Protocol (HTTP, RFC2616). Like SMTP and HTTP, SIP uses Multipurpose Internet Mail Extensions (MIME) to carry extra information. SIP also uses universal resource identifiers (URIs) for addressing in the same way as HTTP.

**Scalability**

Users can be anywhere on the Internet and invited to many different sessions at once.

**Interoperability**

As SIP is an open standard, the development community is able to build a wide variety of implementations that can communicate with other SIP-based products.

SIP permits interaction between devices through signaling messages. These messages can fulfill many purposes, including:

- Registering a user with a system
- Inviting users to join an interactive session
- Negotiating the terms and conditions of a session
- Establishing a media stream between two or more endpoints
- Terminating sessions

### 3.1 Call Scenario using SIP

Fig. 6 shows the steps that SIP setup and release a call.

1. The SIP Phone A sends INVITE message towards SIP Proxy Server, intended for SIP Phone B.

2. The SIP Proxy Server is responsible for receiving a request, determining where to send it based on knowledge of the location of the SIP Phone B, and then sending it there.

3. Both SIP Proxy Server and SIP Phone B respond to INVITE message with Trying(100) message.

4. The Ringing(180) message is sent by SIP Phone B and the SIP Proxy Server relays it.

5. When a user of SIP Phone A hangs up the receiver, the SIP Phone A sends BYE message.

Both SIP Phone A and SIP Phone B send/receive the RTP media flow including voice.

### 4. The Interworking of Megaco/H.248 and SIP

Fig. 7 represents Inter-Networking that Analogue Phone communicates with SIP Phone, using R-MG capable of Megaco/H.248 Protocol. Analogue Phone calls SIP Phone with telephone number of SIP Phone. Also, SIP Phone may call Analogue Phone with telephone number of SIP Phone. So, MGC should translate telephone number of a call and route the call. When SIP Proxy server processes Tel-URL, it uses telephone number translating table loaded with Location Server.

Fig. 8 shows call setup scenario by interworking[4] of Megaco and SIP.
1. The MGC generates the Modify message towards R-MG to check for offhook on the terminations.
2. The Analogue Phone goes off hook.
3. R-MG detects offhook event and sends the Notify message towards the MGC.
4. The MGC issues a Modify command with event descriptor for detecting dial tone and checking digit-map.
5. The R-MG plays Dial Tone call progress tone on analogue interface.
6. The Analogue Phone pushes dialing digits.
7. The R-MG detects the digits dialed and sends the Notify command with dialing digits parameter.
8. The MGC analyzes the dialing digits of destination and find a SIP Proxy for sending SIP INVITE message. After this, The MGC generates SIP INVITE message and sends it to the related SIP Proxy.
9. ~ 11. The SIP Proxy relays SIP INVITE message and also SIP 180 Ringing message of SIP Phone(Called party).
10. The MGC generates a Modify message to the R-MG to apply ringback tone to the given termination.
11. The R-MG after receiving the Modify message applies ringback tone to the Analogue phone.
12. The SIP Proxy relays SIP 200 OK message of SIP Phone(Called party).
13. The MGC meanwhile receives the SIP 200 OK message from the SIP Phone to indicate its SDP information.
14. The MGC generates the Add command for adding the physical termination and to create an ephemeral termination.
15. The R-MG executes the Add command and responds to the command.
16. SIP ACK message is sent toward the SIP Phone(Called party).

5. The Implementation of MG prototype

We implement the MG prototype based on Linux in PC platform. Fig. 11 shows the S/W structure of MG prototype.
analogue line to the Megaco Master and sets events requested by the Megaco Master. The Data Pump manager receives PCM voice data from SLIC/SLAC card, requests the RTP manager to make RTP packet with it, and sends RTP packet returned by the RTP manager towards peer SIP Phone. In this case, The Data Pump manager transfers together codec information for the Codec manager to code PCM voice data to G.732.1 or G.729a. Also, the Data Pump manager receives RTP packet from peer SIP Phone, requests the RTP manager to make PCM voice data with it, and sends the PCM voice data towards SLIC/SLAC card. We use Internet PhoneJack card of the Quicknet Technologies, Inc. as SLIC/SLAC card. Of course, in this card, there are codecs, such as G.711, G.723.1, and G.729a, and DSP for coding/decoding codecs. But, because we hope to test the other codecs, such as AMR-NB(Narrow Band) and SMV, etc, we make Codec manager.

Fig. 11 shows the control flow for controlling MG. The Megaco Slave receives transaction request message from MGC and analyzes it. By the analyzed control information, the Megaco Slave controls the Resource Manager of using the control interface. The Resource Manager plays role of resource management, media(codecs, jitter, etc.) management and connection management. The Resource Manager commands connection(Data_Pump_Request) between channels toward the Data Pump Manager. The Data Pump Manager forwards voice data between POTS interface and IP network.

Fig. 12 shows the environment for testing MGC and MG prototypes. The MG includes SLIC/SLAC card and Analogue Phone is connected with SLIC/SLAC card.

6. Conclusion

As a IP telephony protocol, the SIP is gaining increasing popularity. There is yet POTS(Plain Old Telephone Service) line as the existing subscriber line. The Megaco/H.248 may be used for call between the POTS line and SIP. So, it is needed that method for interworking between Megaco/H.248 and SIP. In this paper, we propose the implementation of Residential Media Gateway Prototype in Interworking between Megaco/H.248 and SIP. We had designed and implemented MG prototype platforms for interworking between POTS line and IP network in interworking of Megaco/H.248 and SIP protocol. And, we had found that it is feasible in the real network.

In the future, we have plan to design and implement the MGC. Also, we have plan to implement data pump manager, RTP manager, and Codec manager as linux kernel modules for increasing performance and interwork between H.323 and Megaco/H.248.

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REFERENCES