JOURNAL OF INFORMATION SCIENCE AND ENGINEERING 23, 787-799 (2007)

An Integrated Call Agent of the Converged VoIP Network

HUNG-HSIN CHANG, MENG-TA HSU, MING-FENG CHANG AND CHIEN-CHAO TSENG
Department of Computer Science and Information Engineering
National Chiao Tung University
Hsinchu, 300 Taiwan

The traditional circuit-switched telecommunication network and the packet-switched data network are converging to a single packet-switched network. One important application of the converged network is telephony communications, also referred to as VoIP (Voice over IP). Call signaling protocols, such as H.323, SIP and MGCP, have been developed to support VoIP communications. To enable devices using different VoIP protocols to communicate, gateways are needed to translate messages of one protocol to messages of another. In this paper, we present a simple, flexible framework for this interworking function. The framework is based on a half-call model where a call is controlled by two half-call finite state machines (FSMs), one representing the state of the caller and the other representing the state of the callee. The interworking function has been implemented such that the caller FSM of one VoIP protocol can interact with the callee FSM of any VoIP protocol. The development effort of the interworking function is minimized since only two half-call FSMs for each VoIP protocol are needed and they can be developed independently as long as the design conforms to the same interface specification. We have developed an integrated call agent (ICA) that contains the half-call FSMs of H.323, SIP and MGCP. Calls between devices using these VoIP protocols can be set up, maintained and terminated by the ICAs.

Keywords: SIP, IN, H.323, MGCP, inter-operation, VoIP

1. INTRODUCTION

The PSTN (Public Switched Telephone Network) has provided reliable voice communication for decades. A telephone call to anyone in the world can be established in seconds, and the voice quality is good in general. Voice waveform transmitted in the PSTN is encoded using PCM (pulse-code modulation, G.711 A-law and u-law, both 64kbps) technique. To establish a telephone call between two parties, a dedicated link needs to be set up, and the link has to be torn down when the call terminates. This work is performed by telephone switches exchanging standard signaling, such as ISUP (Integrated Services Digital Network User Part).

1.1 VoIP Protocols

As the Internet becomes overwhelmingly widespread, transporting voice communication traffic using the Internet Protocol (IP) provides advantages over the traditional PSTN. This is often referred to as IP telephony or VoIP (Voice over IP). VoIP can use sophisticated speech codecs, such as G.723.1 and G.729 [1], to reduce the bandwidth required for a call. VoIP communications use RTP/RTCP to transport voice packets over

Received May 3, 2005; revised July 20, 2005; accepted August 4, 2005.
Communicated by Ten-Hwang Lai.

IP networks [2, 3]. To establish a call, the two parties involved should negotiate the codec and the RTP/RTCP ports used for the call, as well as exchange messages to set up and terminate the call. H.323 and SIP are two existing VoIP signaling protocols. Both are designed to support the setup, communication capacity exchange and tear-down of a VoIP call.

H.323 is an ITU-T recommendation for multimedia conferencing over packet-switched networks [4]. It is a protocol umbrella that consists of many standards, including Q.931 call control protocol, H.225 registration and administration, and H.245 media negotiation. It is widely used nowadays; NetMeeting of Microsoft and VoIP products from Cisco all support H.323 multimedia communications over packet-switched networks. H.323 also defines a gatekeeper as a manager and arbiter over a network. The gatekeeper is an optional entity in charge of endpoint registration, address translation, and bandwidth assignment. An H.323 endpoint can make a direct or gatekeeper-routed call. One of the major criticisms against H.323 is the time and complexity involved in setting up a call; H.323 version 1 uses multiple stages to exchange signaling and media capabilities. Moreover, messages are transported using TCP, which requires additional session set-up time. Recent H.323 versions can include both signaling and media capabilities in a single message transmitted by UDP, which eliminates the additional round-trip time for TCP handshake [5]. However, with too many options, H.323 still has compatibility problems for the products from different providers.

Session Initiation Protocol (SIP) has been standardized by IETF for initiating interactive communication sessions between users [6]. It can be used to establish Internet telephony calls. SIP is a lightweight protocol, which uses only six commands for session control. With early media capability exchange feature, a SIP terminal can issue only one command, INVITE, to set up a call. Therefore, it is faster than H.323 in call establishing. SIP is a text-based request-response architecture similar to HTTP. Since SIP is simpler then H.323, many consider SIP a powerful alternative to H.323.

To enable communications between VoIP users and PSTN users, gateways between the PSTN and IP-based networks are needed. Since the signaling used in the VoIP networks and the PSTN are different, and the voice media are transmitted in different formats, the gateways have to perform two functions: signaling conversion and media conversion. The two functions can be carried out in two separated entities: MGC (Media Gateway Controller) and MG (Media Gateway). An MGC (also referred to as a CA, Call Agent) performs the signaling conversion function, and an MG performs the media conversion function. A standardized protocol can be used between an MGC and MGs. IETF proposed the MGCP (Media Gateway Control Protocol) architecture depicted in Fig. 1 [7]. MGCP is a master-slave protocol. A CA is a master and has control over several MGs; an MG acts as a slave and is kept simple and passive. A CA also performs the call control function as a gatekeeper in H.323, but has much tighter control. A CA instructs an MG to establish, maintain, and terminate a call between a VoIP terminal and an endpoint in the PSTN. MGCP is based on a centralized network infrastructure. To serve to a wide area network, a group of CAs need to coordinate, but MGCP does not specify how the CAs interact. Signaling used in inter-CA communications can be SIP or ISUP. MGCP describes the media capability and parameter using SDP (Session Description Protocol, [8]). Recently Megaco (or H.248), an enhanced version of MGCP, is promoted jointly by the IETF and ITU-T [9].

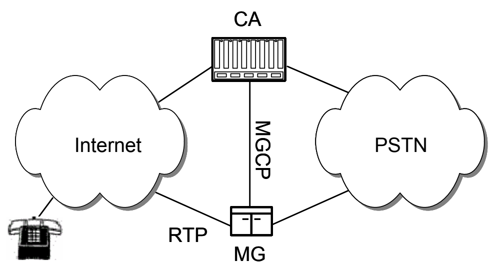


Fig. 1. MGCP architecture.

1.2 Interoperation

Since H.323 and SIP are expected to co-exist in the near future, gateways between H.323 and SIP terminals are needed for the interworking function between H.323 and SIP. EURESCOM proposed a project for providing IN functionality for H.323 telephony calls [10]. Vemuri described an inter-operation model called SPHINX (SIP, H.323 and IN interworking) where H.323 and SIP terminals can access IN services [11]. In addition, an SIP-H323 gateway is a byproduct of this inter-operation model based on the half-call state model of the IN. Agrawal, Schulzrinne and Singh [12, 13] specified the requirements for SIP-H.323 interworking. Gurbani and Rastogi [14] and Haerens [15] suggested ways to map the call control of SIP to IN, but they did not support the H.323 slow-start call setup. Ackermann *et al.*, have implemented a sip-h323 gateway as a basis for supplementary service interworking [16]. They use a dedicated call model that transfers H.323 messages to SIP messages, and vice versa. Singh and Schulzrinne have presented a Columbia InterNet Extensible Multimedia Architecture (CINEMA) where MGCP and H.323 can be interworking through SIP [17]. Nevertheless, it is difficult to modify this call model to cooperate with other VoIP protocols.

However, no work has been done on interoperating all VoIP protocols in a simple and flexible framework. In this paper we present an integrated call agent architecture that supports the interworking function of VoIP protocols (SIP, H.323, MGCP and MEGACO) using the basic call state model in Intelligent Network [18]. The interworking function translates messages of the VOIP protocols. The translation is transparent to the call parties and kept as simple as possible. Furthermore, our design supports inter-CA communications using SIP in a straightforward manner. The rest of the paper is organized as follows. Section 2 briefly describes the IN basic call state model. Section 3 presents the system design and the implementation issues. Implementation issues and results are discussed in section 4. Conclusions are given in section 5.

2. IN BASIC CALL STATE MODEL

An intelligent network (IN) separates the service logic from the switching function in the telecommunications network; the service intelligence is placed in computer nodes that are distributed throughout the network. This provides the network operators with the means to develop and control services more efficiently. New capabilities can be rapidly introduced and customized for the network. A simple IN architecture is depicted in Fig. 2.

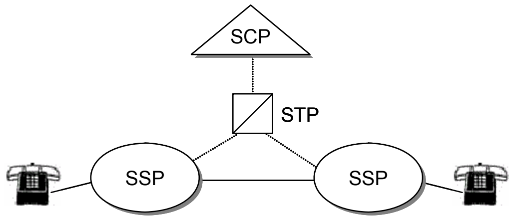


Fig. 2. IN architecture.

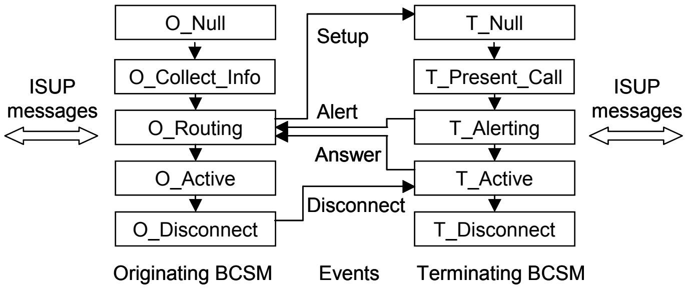


Fig. 3. Simplified IN BCSMs.

A service switching point (SSP) is an IN-capable switching system dealing with the call control functions that establish, maintain, and clear a call. In addition, it detects user requests for IN-based services and queries a service control point (SCP) to determine how the call should be handled. The SCP contains the service logic to provide the IN services.

The basic call control function of an SSP is supported by a finite state machine (FSM) called basic call state model (BCSM). The BCSM consists of point in calls (PICs), detection points (DPs), and events. PICs represent the switching activities or states that a call goes through from call origination to termination. DPs are states at which transfer of control from the SSP to the SCP can occur. Events are messages exchanged between BCSMs, and trigger state transitions of the BCSMs.

Fig. 3 shows that the BCSM is based on half-call model; a call model consists of two half-call models: the originating BCSM (O_BCSM) and terminating BCSM (T_ BCSM). The O_BCSM represents the states associated with the call originating party; the T_BCSM represents the states associated with the call terminating party. Note that the originating and terminating BCSMs interact with the call parties using ISUP messages. The call control functions are performed through the exchange of events between the O_BCSMs and T_BCSMs. These events include Setup, Alert, Answer, Disconnect, Busy, No-Answer, and Abandon. For details, the readers are referred to [18].

3. SYSTEM ARCHITECTURE

We proposed a simple systematic way to implement the interworking function (IWF) between VoIP protocols. There are two major functions in the IWF: call routing function and call control function. The call routing function locates the called party by querying a location database so that the call request can be delivered. How to locate a user is beyond the scope of this paper; we assume that a user location database exists. The call control function sets up and maintains the call by translating messages between two VoIP protocols.

The call control function can be implemented using the BCSMs. For each VoIP protocols, its messages are mapped to the messages of the BCSMs. An interworking gateway of two VoIP protocols can be implemented by combining the BCSMs of the VoIP protocols. Therefore, an O_BCSM of one VoIP protocol and a T_BCSM of another VoIP protocol can interact through the exchange of a set of unified events based on the IN half-call model. Fig. 4 shows the six BCSMs of three major VoIP protocols for a general gateway design. Note that the events between an O_BCSM and a T_BCSM are protocol independent. This design reduces developing efforts. To interwork n different VoIP protocols, each protocol needs only one O_BCSM and one T_BCSM (*i.e.* $2n$ BCSMs in total), instead of hardcoding n^2 messages translation modules between any two protocols.

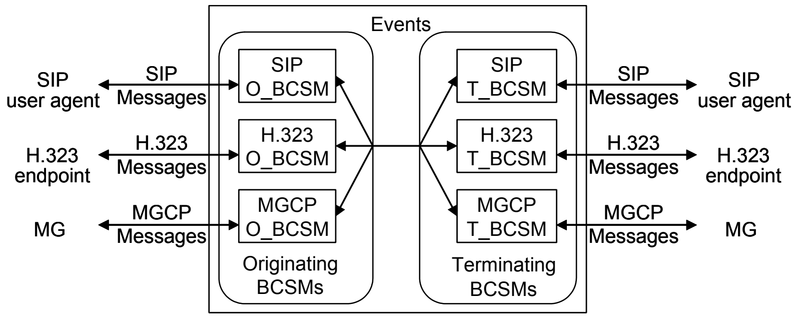


Fig. 4. Components developed for a general VoIP gateway.

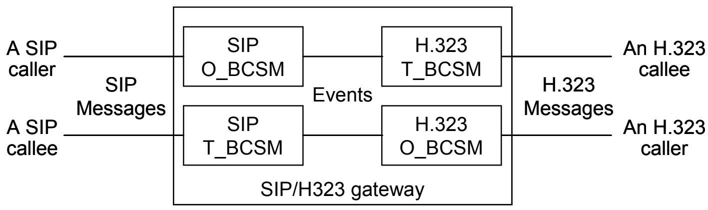


Fig. 5. A SIP/H.323 gateway.

A combination of the BCSMs of two VoIP protocols becomes a gateway; the construction of a gateway is to design the BCSMs of the VoIP protocols. Fig. 5 shows an example SIP/H.323 gateway. This gateway consists of the BCSMs of SIP and H.323. When a SIP UA originates a call to a H.323 terminal, a SIP O_BCSM receives a SIP INVITE message and issues a Setup event to an H.323 T_BCSM. After being notified by the Setup event, the H.323 T_BCSM sends an H.323 SETUP message to the terminating H.323 endpoint. Subsequent SIP and H.323 messages are exchanged to set up the call through the interaction of the SIP O_BCSM and H.323 T_BCSM. On the other hand, an originating call from a H.323 terminal to a SIP UA can be handled by an H.323 O_BCSM and a SIP T_BCSM.

All the gateways needed in the converged VoIP network can be implemented in the same way described above. Fig. 6 depicts a converged network using three gateways: SIP/H.323, SIP/MGCP, and MGCP/H.323 gateways. Note that the gateways can share the same BCSMs developed. Moreover, when a new VoIP protocol, such as MEGACO, is added to the converged network, the gateways to MEGACO network use the same BCSMs developed for existing VoIP protocols; only the BCSMs of MEGACO need to be implemented.

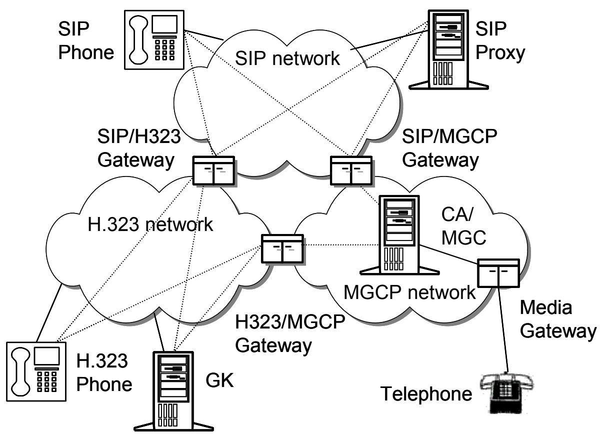


Fig. 6. A converged VoIP network using gateways.

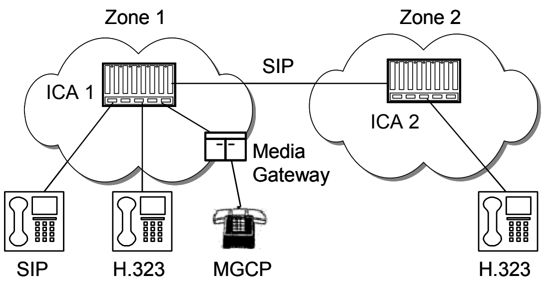


Fig. 7. A converged VoIP network managed by integrated call agents.

To route an originating call, the location of the callee and the signaling protocol that the callee supports must be determined so that a correct T_BCSM can be invoked for messages translation. Solutions to this call routing function are beyond the scope of this paper; we assume that a solution is available. We integrate the call routing function and the BCSMs of all VoIP protocols in an entity called integrated call agent (ICA). The ICA can not only translate messages between VoIP protocols but also act as a H.323 gatekeeper, a SIP proxy/registrar, and a MGCP call agent.

The design is extensible. Fig. 7 shows the converged architecture partitioned into two zones managed by two ICAs. Calls between entities of two zones can be set up by the ICAs exchanging SIP messages through two SIP BCSMs. Fig. 8 depicts an example call between an H.323 terminal and an MGCP phone, and two ICAs (ICA 1 and ICA 2) are involved. When an H.323 terminal in Zone 1 initiates a call to an MGCP phone in Zone 2, ICA 1 invokes a SIP T_BCSM and sends a SIP INVITE message to ICA 2. ICA 2 initializes a SIP O_BCSM to handle this call request. Since ICA 2 knows that the callee is an MGCP phone, an MGCP T_BCSM was invoked to set up this call connection.

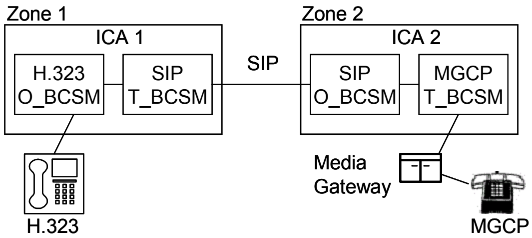


Fig. 8. An example of H.323 and MGCP interworking using 2 ICAs.

3.1 Mapping of VoIP Protocol Messages to the BCSM Messages

The implementation of half-call BCSM for each VoIP protocol is to map the VoIP call signaling messages to the IN BCSM messages. We focus on the relationship between the BCSM states and the VoIP messages. This message mapping is straightforward except for the slow-start version of H.323. Fig. 9 depicts the mapping for H.323, SIP, and MGCP.

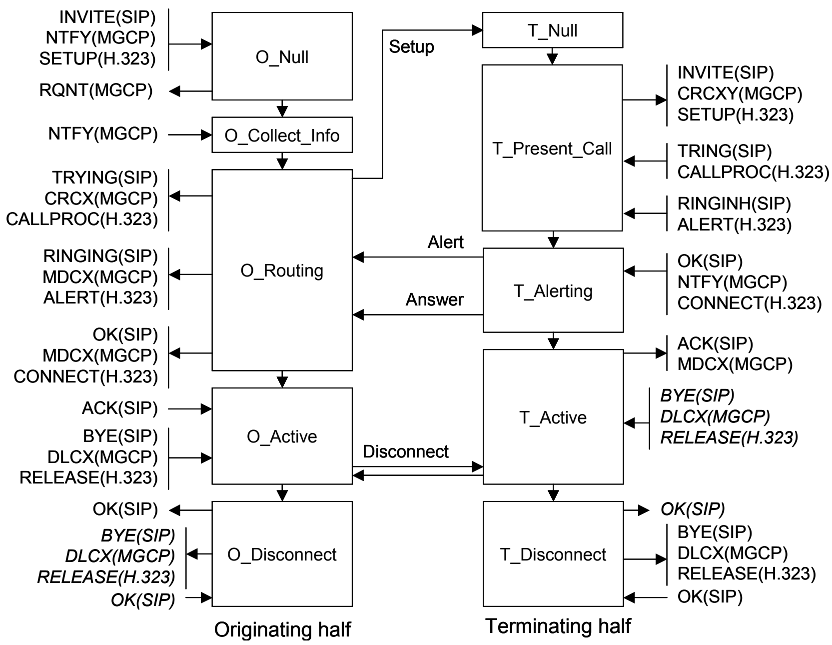


Fig. 9. Mapping VoIP messages to BCSM messages.

When a call request message arrives (*e.g.*, INVITE for SIP, SETUP for H.323, or NOTIFY for MGCP) from a calling party, the corresponding O_BCSM becomes active and initialize itself to state *O_Null*. The O_BCSM proceeds to state *O_Collect_Info* where a called number or address is collected from the calling party. For SIP and H.323, the INVITE and SETUP messages carry the identifier of the called party, media descriptor, and signaling transport address. Hence the O_BCSM simply skips state *O_Collect_ Info* and go to state *O_Routing* directly. For MGCP, however, a RQNT message should be sent to the calling party to collect the dialed number. Once the calling party collects the number dialed, it sends a NTFY with the dialed number to the O_BCSM. Then, the O_BCSM transits to state *O_Routing*.

State *O_Routing* responds to the calling party with the call progressing message (*e.g.*, 100 Trying for SIP or CALLPROC for H.323). In addition, the VoIP protocol used by the called party is determined, and a Setup event is issued. Consequently, a T_BCSM for the called party is initialized at state *T_Null* to handle this Setup event. The O_BCSM stays in state *O_Routing* waiting for the Alert and Answer events sent from the T_BCSM. If both events are received, the call has been accepted by the called party and the O_BCSM responds with a call connect message (such as CONNECT for H.323, OK for SIP, or MDCX for MGCP) to the calling party indicating that the call has been accepted. The O_BCSM stays at state *O_Active* until the call is terminated.

When a T_BCSM is activated by a Setup event from an O_BCSM, it goes to state *T_Present_Call* and checks the validation of the called number. If the number is valid, the T_BCSM sends a call request message (*e.g.*, INVITE for SIP, SETUP for H.323, or CRCX for MGCP) to the called party and waits for the response messages such as trying, altering, and answered from the called party. Upon receiving the alerting message (*e.g.*, 180 Ringing for SIP or ALERT for H.323), the T_BCSM informs the O_BCSM with an Alert event and proceeds to state *T_Alert*.

In state *T_Alert*, the T_BCSM waits for the call connect message (*e.g.*, OK for SIP, CONNECT for H.323, or NTFY for MGCP). Once the message is received, the call has been accepted by the called party. The T_BCSM activates an Answer event to the O_BCSM and transits to state *T_Active*. In this state, an acknowledgement message (*e.g.* ACK for SIP or MDCX for MGCP) is sent back to the called party. Thus far, a basic two party call connection is established.

If a disconnect message (*e.g.*, BYE for SIP, RELEASE for H.323, or DLCX for MGCP) is received by either the originating or the terminating BCSM, the BCSM will issue a Disconnect event to the corresponding BCSM of this call. Both BCSMs proceeds to state *Disconnect*, and the call is terminated.

3.2 H.323 Slow-Start

For the H.323 slow-start, the mapping is quite different and those events described above are inadequate to exchange the media capability. For the BCSMs described above, after the media capability of the calling party becomes available at state *O_Null*, the O_BCSM issues a Setup event. Moreover, after the capability of the called party becomes available at state *T_Present_Call*, the T_BCSM issues an Answer event respectively. However, for H.323 slow-start, this capabilities will not be determined until the H.245 negotiation after the CONNECT messages. As a result, the BCSMs do not get

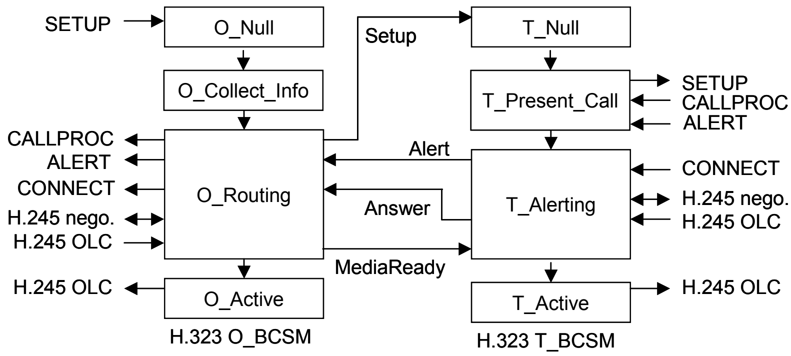


Fig. 10. BCSMs for the H.323 slow-start.

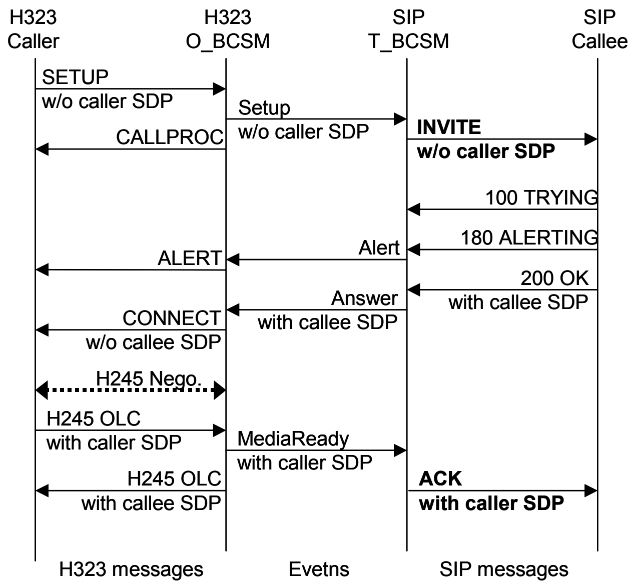


Fig. 11. Call flow of H.323 (slow-start) and SIP interworking.

media capability when the Setup or Answer event is issued. In addition, there is no event reporting the media capability to the calling and called party after the Answer event. Thus, we made two modifications to the H.323 BCSMs. First, the Answer event is postponed to state *T_Alert* where the T_BCSM receives an H.245 open logical channel message (OLC) which carries the media capability for the called party. Second, a new event, media capability ready (MediaReady), is introduced at the end of state *O_Routing* to notify the T_BCSM the media capability of the calling party. Fig. 10 shows the modifications to the mapping for the H.323 slow-start.

To interwork with the H.323 BCSMs that supports slow-start feature, the BCSMs of SIP and MGCP were modified to send a MediaReady event anyway at the end of state *O_Routing*. *i.e.*, all O_BCSMs should issue a MediaReady event at state *O_Routing*. Although the called party of SIP or MGCP expects to receive the media capability when it receives the INVITE or the CRCX message, this capability exchange can be delayed till when the ACK or the MDCX message is received. Fig. 11 shows the call flow of H.323 and SIP interworking where the H.323 terminal is in slow-start mode. The call flow of a slow-start H.323 terminal and an MGCP phone can be done in a similar manner. If the calling party is an H.323 phone using the slow-start feature, the T_BCSM issues an INVITE message without specifying the caller's media capability to the SIP callee. The media capability is available at state *O_Routing* after the H.245 OLC message is received. Consequently, the O_BCSM issues a MediaReady event to the T_BCSM at the state *O_Routing*. Therefore, the T_BCSM encloses this capability in the ACK message sent to the called party.

4. SYSTEM IMPLEMENTATION AND RESULTS

To reduce the effort in developing a VoIP gateway, we use existing, well-developed protocol stacks and endpoints. In our implementation, we use the MGCP and SIP protocol stacks developed by CCL/ITRI, Taiwan and the open-source H.323 protocol stack developed by OpenH323. In addition, our experiment platform includes two residential gateways (RGWs) which are also developed by ITRI using D/41E and D/41ESC cards from Dialogic Corp. The RGWs support MGCP and can connect up to 16 telephones. The ICA platform also supports Microsoft NetMeeting (using H.323) and SIP user agent. The H.323 BCSMs are modified from the source code of the OpenH323's OpenGate that supports registration administration status (RAS) messages and gatekeeper-routed call signaling. We have also developed a SIP proxy/registrar based on the ITRI SIP protocol stack. Fig. 12 summarizes the components used in our platform.

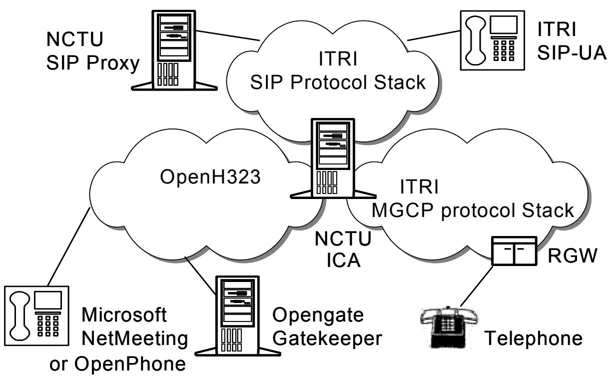
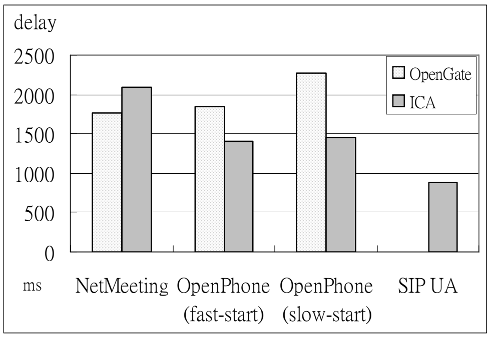


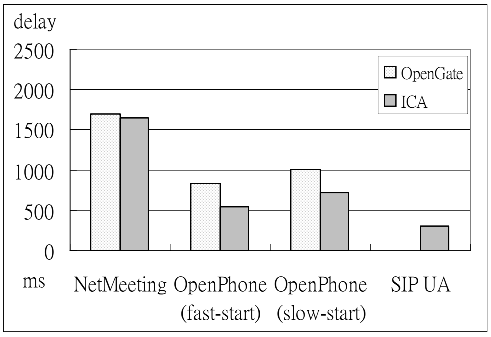
Fig. 12. Components used in our platform.

In our experiments, a call can be successfully set up between any two VoIP phones. The Microsoft NetMeeting currently supports only H.323 slow-start version; we use an open-source OpenPhone (with both slow-start and fast-start capabilities) to test the cases of slow-start version. In addition, a Vocal sip proxy, developed by Vovida, was used to test calls between SIP UAs. Since an ICA acts as both SIP proxy and H.323 gatekeeper, an ICA can initiate a call to the phones that are under the control of a SIP proxy or an H.323 gatekeeper without further modification.

The comparison of the delays in establishing a call between various types of phones by OpenGate H.323 gateway, our ICA, and Vocal SIP proxy is depicted at Fig. 13. No result of inter-protocol calls through OpenGate and Vocal is listed, because they do not convert the messages of different protocols. Fig. 13 (a) shows call establishment delays for the calls initiated from various types of phones to NetMeeting, which only equipped with slow-start mode, and Fig. 13 (b) shows those for calls to OpenPhone in H.323 fast-start Mode. Although our ICA supports signaling conversion for different VoIP protocols, the results indicates that the ICA sets up calls faster than the OpenGate in all cases except for calls between two NetMeeting users.



(a) Calls to NetMeeting (slow-start).



(b) Calls to OpenPhone (fast-start).

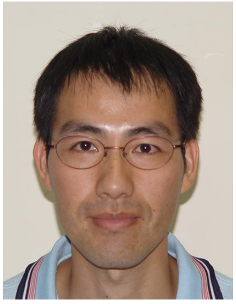
Fig. 13. Call establishment delays.

5. CONCLUSIONS

We have presented a simple, flexible framework for the interworking functions of VoIP protocols based on IN half-call BCSM. In addition, we have implemented the basic gateway components, O_BCSMs and T_BCSMs, for SIP, H.323, and MGCP. Using these components, gateways for SIP/H.323, SIP/MGCP, and MGCP/H.323 can be constructed. This approach simplifies the effort in interworking with a call signaling protocol, such as ISUP and Q.931, in the network. By using the same interaction events of the half-call model, the BCSMs of a call signaling protocol can interwork with the existing BCSMs. In addition, an ICA containing all the BCSMs is able to translate messages between call signaling protocols. Under this half-call control framework, a converged VoIP network can be managed by a group of coordinating ICAs such that two user devices managed by different ICAs can communicate. The call routing function that determines the location and protocol of the called party has not been fully investigated in this paper. As a mobile user may change his IP address and VoIP devices constantly, this problem becomes even more complicated. We need registration and/or paging schemes to track mobile users in the converged telecommunication network. Recently, P2P (peer-to-peer) VoIP communications, such as Skype, have become very popular. The interworking function for a P2P VoIP system and a client-server one (such as SIP) is an important issue that needs to be investigated.

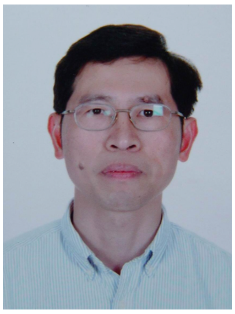
REFERENCES

1. ITU, "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-CELP)," *Recommendation G.729*, ITU-T.1, Geneva, 1996.
2. S. Casner, R. Frederick, V. Jacobson, and H. Schulzrinne, "RTP: a transport protocol for real-time applications," IETF Request for Comments 1889, 1996.
3. H. Schulzrinne and L. C. Stephen, "RTP profile for audio and video conferences with minimal control," IETF Request for Comments 1890, 1996.
4. ITU, "Packet-based multimedia communication systems," *ITU-T Recommendation H.323*, 1998.
5. ITU, "Call signaling over UDP," *ITU-T Recommendation H.323 Annex E*, 1998.
6. G. Camarillo, A. Johnston, M. Handley, J. Peterson, J. H. Rosenberg, E. S. Schooler, and R. Sparks, "SIP: session initiation protocol," IETF Request for Comments 3261I, 2002.
7. M. Arango, A. Dugan, I. Elliott, C. Huitema, and S. Pickett, "Media gateway control protocol (MGCP) version 1.0," IETF Request for Comments 2705, 1999.
8. M. Handley and V. Jacobson, "SDP: session description protocol," IETF Request for Comments 2327, 1998.
9. C. Groves and M. Pantaleo, "The megaco/H.248v2 gateway control protocol, version 2," IETF Internet Draft draft-ietf-megaco-h248v2-03.txt, 2002.
10. EURESCOM, "Providing IN functionality for H.323 telephony calls," Project Report P916, Research and Strategic Studies in Telecommunications, European Institute, 1999.
11. K. V. Vemuri, "SPHINX: a study in convergent telephony," in *Proceedings of the IP Telecom Services Workshop*, 2000, pp. 9-18.
12. K. Singh and H. Schulzrinne, "Interworking between SIP/SDP and H.323," in *Proceedings of the 1st IP-Telephony Workshop*, 2000, pp. 75-92.
13. H. Agrawal, V. Palawat, and R. R. Roy, "SIP-H.323 interworking," IETF Internet Draft draft-agrawal-sip-h323-interworking-reqs-02.txt, 2001.
14. V. K. Gurbani and V. Rastogi, "Accessing IN services from SIP networks," IETF Internet Draft draft-gurbani-iptel-sip-to-in-05.txt, 2001.
15. F. Haernes, "SIP-IN interworking protocol architecture and procedures," IETF Internet Draft draft-haerens-sip-in-00.txt, 2001.
16. R. Ackermann, V. Darlagiannis, M. Goertz, M. Karsten, and R. Steinmetz, "An open source H.323/SIP gateway as basis for supplementary service interworking," in *Proceedings of the 2nd IP Telephony Workshop*, 2001, pp. 1-7.
17. W. Jiang, J. Lennox, S. Narayanan, H. Schulzrinne, and K. Singh, "CINEMA: columbia InterNet extensible multimedia architecture," Technical Report CUCS-011-02, Department of Computer Science, University of Columbia, 2002.
18. U. Black, *The Intelligent Network: Customizing Telecommunication Networks and Services*, Prentice Hall, New Jersey, 1998.



Hung-Hsin Chang (張弘鑫) received the B.S. and M.S. degree in Computer Science and Information Engineering from the National Chiao Tung University, Taiwan, R.O.C., in 1990 and 1995. He is currently working toward the Ph.D. degree in Computer Science and Information Engineering from the National Chiao Tung University. His current research interests include design and analysis of wireless network, Internet communications, and personal communications network.

Meng-Ta Hsu (許孟達) received the B.S. degree in Computer Science and Information Engineering from the National Taiwan University, Taiwan, R.O.C., in 2000. He is currently working toward the Ph.D. degree in Computer Science and Information Engineering from the National Chiao Tung University. His current research interests include Internet communications, intelligent transportation systems, mobile computing, and performance modeling.



Ming-Feng Chang (張明峰) received the Ph.D. degree in Computer Science from the University of Illinois at Urbana-Champaign in 1991. He is currently a Professor in the Department of Computer Science and Information Engineering, National Chiao Tung University, Taiwan, R.O.C. His current research interests include design and analysis of Internet communications, personal communications network, mobile payment, and performance modeling.

Chien-Chao Tseng (曾建超) is currently a professor in the Department of Computer Science at National Chiao Tung University, Hsinchu, Taiwan. He received his B.S. degree in Industrial Engineering from National Tsing Hua University, Hsinchu, Taiwan, in 1981; M.S. and Ph.D. degrees in Computer Science from the Southern Methodist University, Dallas, Texas, U.S.A., in 1986 and 1989, respectively. His research interests include wireless Internet, handover techniques for heterogeneous networks, and mobile computing.