

The Behavior of IAX Protocol: A Mathematical Analysis

Hadeel Saleh Haj Aliwi[†] and Putra Sumari^{††},

School of Computer Sciences, Universiti Sains Malaysia, Penang, Malaysia

Summary

Over the last few years, many multimedia conferencing and Voice over Internet Protocol (VoIP) applications have been developed due to the use of signaling protocols in providing video, audio and text chatting services between at least two participants. This paper studies the behavior of the widely common signaling protocol; InterAsterisk eXchange Protocol (IAX) in terms of the behavior of the signaling and media messages, as well as the delay time during call setup, call teardown, and media sessions.

Key words:

Signaling Protocol, Media Conferencing, VoIP, IAX

1. Introduction

With the appearance of numerous multimedia conferencing and Voice over Internet protocols [1,2,3], the decision to choose the appropriate protocol to be utilized in such a service has become very difficult since each protocol has its own privileges which differ from the corresponding privileges of the other protocols.

Choosing IAX protocol to be discussed is due to many reasons; IAX is an interesting alternative compared to the conventional VoIP protocols. Nowadays, IAX is being deployed by service providers for their VoIP service offerings (e.g. H.323 and SIP). IAX protocol offers significant features that are not provided by other existent VoIP signaling protocols. Furthermore, many researchers have shown that IAX is slightly better than SIP [4,5], H.323 [6], MGCP [7] and RSW [9,10] in terms of quality of services.

2. IAX Protocol

In 2004, Mark Spencer has created the Inter-Asterisk eXchange (IAX) protocol for asterisk that performs VoIP signaling [10]. IAX is supported by a few other softswitches, (Asterisk Private Branch eXchange) PBX systems [11], and softphones [12]. Any type of media (Video, audio, and document conferencing) can be managed, controlled and transmitted through the Internet Protocol (IP) networks based on IAX protocol [13]. IAX2 is considered to be the current version of IAX. The IAX's first version is obsolete. User Datagram Protocol (UDP)

[14,15] is the only protocol that is used by either IAX2 or IAX as a transport protocol. More specifically, UDP is usually used on port 4569 where port 5036 is used by IAX1. The same UDP port is used throughout media transmission and signaling information sessions. IAX mini and full frames are used to carry the media packets during the call [16].

IAX supports the trunk connections concept for numerous calls. The bandwidth usage is reduced when this concept is being used because all the protocol overhead is shared by two IAX nodes for the whole calls. Over a single link, IAX provides multiplexing channels [17,18].

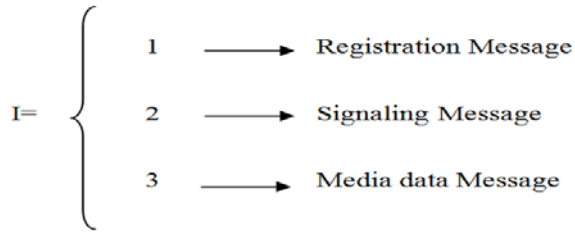
IAX protocol has three main procedures used for the media conferencing between two IAX endpoints, which are call setup, media transmission, and call teardown with the steps of each procedure.

Endpoint A Sends NEW packet to the Endpoint B to place a call, and wait until receiving ACCEPT packet from Endpoint B. After ACCEPT reply, Endpoint A sends ACK packet to Endpoint B to acknowledge of receiving the ACCEPT packet by Endpoint A. After that, Endpoint B rings at Endpoint A by sending RINGING packet, which in turn send ACK packet to Endpoint B to inform about receiving ACCEPT message. Then, Endpoint B sends ANSWER packet to Endpoint A in order to start the call, and wait till sending the acknowledgment message (ACK) by endpoint A.

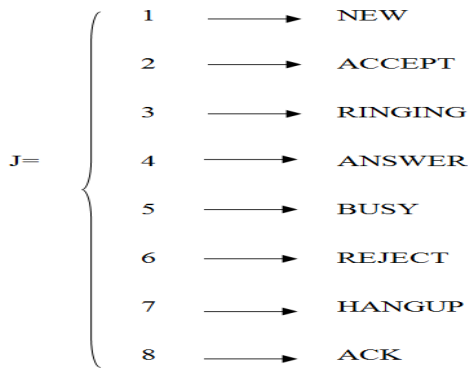
At that time, the audio conferencing is started by transferring the audio packets between the two endpoints which is carried by the IAX mini and full frames. Once the two endpoints complete their call, Endpoint A sends HANGUP packets to Endpoint B to end the call, finally Endpoint B do reply back by sending acknowledgment packet (ACK).

3. IAX Messages Analysis

Each control protocol has three types of sessions; registration session, call setup and teardown sessions, and media data session. In this section each protocol session type is presented by number I varies from 1 to 3 as number as the session types, so when i=1 means the current message is related to the registration session, and same when i=2 and 3. In this paper, only the behavior of the signaling and media data messages will be discussed.



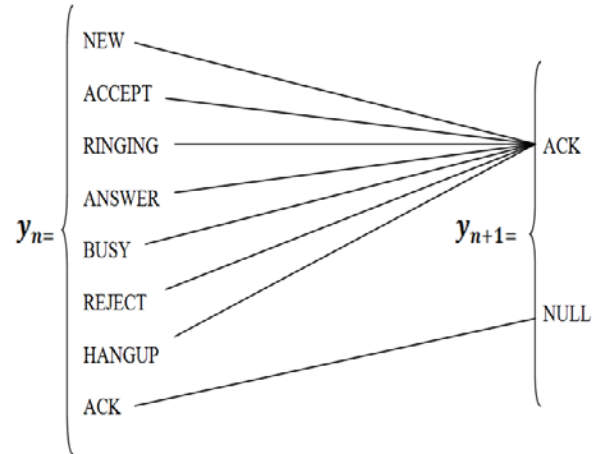
Each one of the aforementioned messages has lists of messages type related to a certain session. For example, the signaling messages can be either initiate message or ringing message or accept message or terminate message. IAX signaling message type is presented by number j varies from 1 to 8 as number as the signaling messages.



As the protocol message type is presented by I , so as an example when $i=2$ and $j=5$ means the message is related to the IAX signaling session which is BUSY. As a result, to identify exactly each protocol message belongs to which session and what is the message type during that session, each IAX message can be presented by I which is the number of the session and J which is the number of the message inside the session i .

$$\text{IAX Message Type} = M_{ij} \quad (1)$$

During the IAX signaling session, if client 1 sends message to client 2 (except acknowledgement message), the latest should notify client 1 of receiving the message successfully before receiving the next message. Otherwise, no message has to be sent back to client 1. If the IAX message sent from client 1 to client 2 is presented by the function y_n , so the notification message sent back to client 1 is presented by the function y_{n+1} . With more details:



As each IAX message is defined by the number of session and the number of the message inside the session, therefore:

$$y_n = \left\{ \begin{array}{ll} \bigvee_{j=1}^7 M_{2j} & \longrightarrow M_{28} \\ M_{28} & \longrightarrow \text{NULL} \end{array} \right. \quad y_{n+1} =$$

During the call setup, the caller sends an acknowledgement message to the callee three times, firstly when the caller is informed about the acceptance of the negotiation by the callee, secondly when the caller is informed about the ringing process by the callee, thirdly when the caller is informed that the callee answers the call. Thus, the status of the call can be identified by counting the number of the acknowledgement messages during the signaling session.

$$\text{Status} = \left\{ \begin{array}{ll} \text{The callee is busy/ the call is rejected} & \text{when } N_{ACK} = 1 \\ \text{The callee is available but he/she does not answer the call} & \text{when } N_{ACK} = 2 \\ \text{The callee answers the call} & \text{when } N_{ACK} = 3 \end{array} \right.$$

In media session, j represents the order of a certain media message. J varies from 1 to r where 1 represents the order of the first IAX message, r is an integer positive number represents the order of the last IAX message respectively. Assuming that N is the number of messages, so number of IAX media messages = (the order of the last media message – the order of the first media message) + 1. Thus,

$$N_{M_{3j}} = (r - 1) + 1 = r \quad (2)$$

Since the payload of IAX client carried by mini header in the media session, by assuming that payload is presented by PL, so

$$Mini_{PL} = M_{3j} \quad (3)$$

4. IAX Sessions Time Analysis

In IAX Environment, two main sessions has to be considered which are signaling session and media session. Signaling session is divided into two sessions; setup session and teardown session. The delay time of each message is the difference between the message's sending time from client 1 and the message's receiving time by client 2. In order to calculate the time spent to complete the call setup session, the time difference between the first IAX message's sent by Client 1 and the last IAX message received by client 2 during the setup session should be measured. Assuming that T presents the message time, T_S presents the sent time of the message, and T_R presents the received time of the message.

$$T_{M_{2j}} = T_{R_{M_{2j}}} - T_{S_{M_{2j}}} \quad (4)$$

$$T_{Setup_{IAX}} = T_{R_{ACK}} - T_{S_{NEW}} \quad (5)$$

As there are four ACK messages during the call, one after initiating the call and getting the negotiation acceptance, one after the ringing message, one once answering the call, and one after terminating the call. Each ACK message has different time that the other message, so to differentiate between the five messages, each ACK message has to be defined by a number q varies from 1 to 3.

$$ACK = M_{28q} \quad \text{where } q=1, 2, 3.$$

Hence,

$$T_{Setup_{IAX}} = T_{R_{M_{283}}} - T_{S_{M_{21}}} \quad (6)$$

In order to calculate the time spent to complete the call teardown session, the time difference between HANGUP message sending time and ACK message arrival time should be calculated.

$$T_{Teardown_{IAX}} = T_{R_{ACK}} - T_{S_{HANGUP}} \quad (7)$$

Hence,

$$T_{Teardown_{IAX}} = T_{R_{M_{284}}} - T_{S_{M_{27}}} \quad (8)$$

Similar to the signaling message sending/receiving status, the delay time of each media message is the difference between its sent and arrival times.

$$T_{M_{3j}} = T_{R_{M_{3j}}} - T_{S_{M_{3j}}} \quad (9)$$

In order to calculate the time spent to complete the media session, the time difference between HANGUP message sending time and ACK message arrival time (once the call answered) has to be calculated.

$$T_{Media_{IAX}} = T_{S_{M_{27}}} - T_{R_{M_{283}}} \quad (10)$$

5. Conclusion

This paper provided a mathematical analysis of the behavior of the IAX protocol in terms of the signaling and media messages, and the time of each call setup, call teardown, and media sessions.

References

- [1] Goode, B: Voice over internet protocol (VoIP), Proceedings of the IEEE, 1495–1517, 2002.
- [2] Haj Aliwi, H. S. and Sumari P.: A Comparative Study of VoIP Protocols, International Journal of Computer Science and Information Security (IJCSIS), USA, 11(4), 97-101, 2013.
- [3] Haj Aliwi, H. S. and Sumari P.: A Comparative Study of VoIP, MCS, Instant Messaging Protocol and Multimedia Applications, (IJCSNS) International Journal of Advances in Computer Networks and Its Security, 67-70, 2014.
- [4] Geneiatakis, D., Dagiuklas, T., Kambourakis, G., Lambrinouidakis, C. and Gritzalis, S.: Survey of security vulnerabilities in session initial protocol, IEEE Communications Surveys & Tutorials, 8(3), 68-81, 2006.
- [5] Glasmann, J., Kellerer, W., and Muller, H.: Service Architectures in H.323 and SIP: A Comparison, IEEE Communications Surveys & Tutorials, 5(2), 32-47, 2003.
- [6] Basicovic, I., Popovic, M. and Kukulj, D.: Comparison of sip and h.323 Protocols, The Third International Conference on Digital Telecommunications ICDT '08, Bucharest, 162–167, 2008.
- [7] Dajiuklas, T., Ioannou, K., and Garmpis, A.: A Lightweight and Scalable VoIP platform based on MGCP/H.323 interworking and QOS management capabilities, Proceedings of the 4th WSEAS International Conference on Information Security, Communications and Computers, Tenerife, Spain, 548-553, 2005.
- [8] Haj Aliwi, H. S., Alomari, S. A., and Sumari P.: An Efficient Audio Translation Approach between SIP and RSW Protocols, Proceedings of 3rd World Conference on

- Information Technology (WCIT-2012), University of Barcelona, Barcelona, Spain, 31-37, 2013.
- [9] Ramadass, S., Subramanian, R. K., Guyennet, H., and Trehel, M.: Using RSW Control Criteria to Create a Distributed Environment for Multimedia Conferencing, Proceedings of the Research and Development in Computer Science and its Applications, Penang, Malaysia, 27-29, 1997.
 - [10] Spencer, M., Capouch, B., Guy, E., Miller, F., and Shumard, K.: IAX: Inter-Asterisk eXchange Version 2, <http://tools.ietf.org/html/rfc5456>, 2010.
 - [11] Spencer, M.: Configuring IAX Clients, <http://www.voip-info.org/wiki/view/Asterisk+config+iax.conf>, 2004.
 - [12] Reeves, E.: Testing Devices That Handle Inter-Asterisk eXchange, Version 2 (IAX2) Protocol, <http://blogs.ixiacom.com/ixia-blog/content-aware-testing-iax2-protocol/>, 2011.
 - [13] Spencer, M. and Miller, F. W. IAX Protocol Description, <http://www.voipinf.hu/content/download/iax.pdf>, 2004.
 - [14] Dinicolo, D.: Transporting VoIP Traffic with UDP and RTP, <http://www.2000trainers.com/voip/voip-udp-rtp-protocol/>, 2007.
 - [15] Forouzan, B. Data Communications and Networking, 4th edition, McGrawHill, New York, USA, 2007.
 - [16] Haj Aliwi, H. S. and Sumari P.: Real Time Audio Translation Module between IAX and RSW, (IJCNC) International Journal of Computer Networks & Communications, 125-133, 2014
 - [17] Kolhar, M., Abu-Alhaj, M., Abouabdalla, O., Wan, T.C., and Manasrah, A.: Comparative Evaluation and Analysis of IAX and RSW, International Journal of Computer Science and Information Technology (IJCSIS), USA, 6(3), 250–252, 2013.
 - [18] Haj Aliwi, H. S., Sumari, P., Alomari, S. A.: An Efficient Interworking between Heterogeneous Networks Protocols and Multimedia Computing Applications, International Journal of Computer Science and Information Security (IJCSIS), USA, 11(5), 81-86, 2013.

Hadeel Saleh Haj Aliwi received her Bachelor degree in Computer Engineering from Ittihad Private University, Syria in 2007-2008 and Master degree in Computer Science from Universiti Sains Malaysia, Penang, Malaysia in 2011. Currently, she is a PhD candidate at the School of Computer Science, Universiti Sains Malaysia. Her main research area interests are in includes Multimedia Networking, VoIP protocols, Interworking between Heterogeneous protocols, and Instant Messaging protocols.

Putra Sumari obtained his MSc and PhD in 1997 and 2000 from Liverpool University, England. Currently, he is Associate Professor and a lecturer at the School of Computer Science, USM. He is the head of the Multimedia Computing Research Group, CS, USM. Member of ACM and IEEE, Program Committee and reviewer of several International Conference on Information and Communication Technology (ICT), Committee of Malaysian ISO Standard Working Group on Software Engineering Practice, Chairman of Industrial Training Program, School of Computer Science, USM, Advisor of Master in Multimedia Education Program, UPSI, Perak