An Analytical Model for the Behavior of SIP, RSW, and H.323 Messages and Session Time

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Abstract: - Over the last few years, many multimedia conferencing and Voice over Internet Protocol (VoIP) applications have been developed due to the use of signalling protocols in providing all types of chatting services between at least two participants. This paper compares between the behaviours of each of the widely used common signalling protocols; H.323 Protocol, Session Initiation Protocol (SIP), and Real-time SWitching Control Protocol in terms of the behaviour of the signalling and media messages, as well as the delay time during call setup, call teardown, and media sessions.

Key-Words: - VoIP, Signaling Protocol, SIP, RSW, H.323, Analytical Model

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.

Three protocols have been chosen to be compared in terms of their messages behaviours which are SIP, RSW, and H.323. Choosing the three protocols to be discussed in this paper is due to many reasons; they being deployed by service providers for their VoIP service offerings, as well as they offer significant features that are not provided by other signalling protocols.

2 Background

2.1 SIP Protocol

SIP is an application-layer control protocol [4] [5] that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls [6].

SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established. SIP does not carry any voice or video data itself. It merely allows two endpoints to set up connection to transfer that traffic between each other via Real-time Transport Protocol (RTP) [7]

User Datagram Protocol (UDP) and Transport Control Protocol (TCP) [7] are transport protocols used to transfer audio and video data. SIP uses many signalling messages in order to handle the communication between two nodes or more. SIP makes use of the request methods; INVITE, ACK, OPTIONS, BYE, CANCEL, etc in order to control the call setup and call teardown [8] [9].

2.2 RSW Protocol

Real-time Switching (RSW) control criteria is a control protocol used to handle a multipoint-to-multipoint multimedia conferencing sessions [10] [11]. RSW control protocol was developed in 1993 as a control mechanism for conferencing by the Network Research Group in school of computer sciences, University of Sciences Malaysia (USM) [12].

RTP protocol is used by RSW control protocol to carry audio and video data through multimedia conferencing [13]. UDP transport protocol used by RSW to transfer audio and video data. RSW control criteria is involved in decreasing bandwidth when many clients using the MCS system. RSW makes a list of priority for the participants to avoid confusion when many participants are trying to speak up during conference [14] [15].

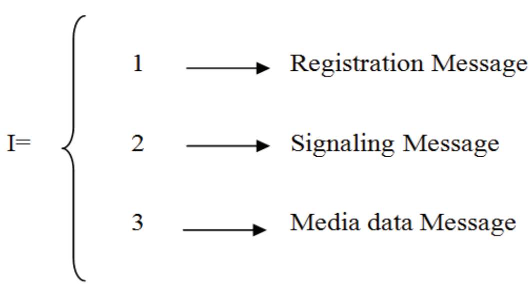
2.3 H.323 Protocol

H.323 is an umbrella standard that provides well-defined system architecture [16] [17], and implementation guidelines that cover call set-up, call control, and the media used in the call [18]. It was established by the International Telecommunications Union (ITU) as the first communications protocol for real time multimedia communication over IP. H.323 takes the more telecommunications-oriented approach to voice/video over IP.

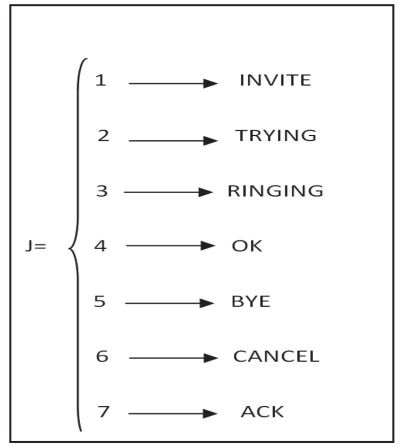
H.323 protocol provides a comparable functionality using different mechanisms and offers highly network management and interoperability [19]. H.323 protocol uses either TCP or UDP to transmit the audio/video packet to the destination side. As well as, RTP is used to carry the media packets via Internet.

3 Signaling Protocols' Messages Analysis

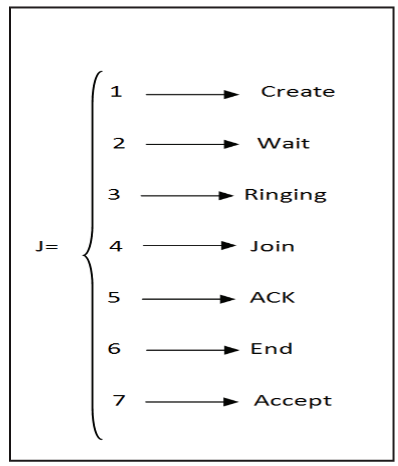
Each control protocol has three types of sessions; registration session, call setup and teardown sessions, and media data session. In this section each protocol session type is presented by number I varies from 1 to 3 as number as the session types, so when i=1 means the current message is related to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



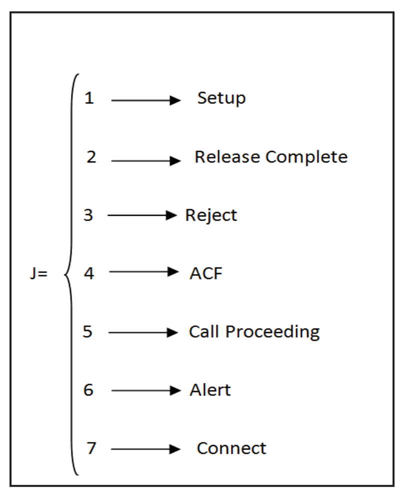
Each one of the aforementioned messages has lists of messages type related to a certain session. For example, the signalling messages can be either initiate message or ringing message or accept message or terminate message. SIP Signalling message type is presented by number j varies from 1 to 7 as number as the signalling messages. Also, both RSW and H.323 signalling message types include 7 messages each.



SIP Signalling Messages



RSW Signalling Messages



H.323 Signalling 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 SIP signalling session which is BYE or the message is ACK in case of RSW protocol and Call Proceeding message in case of H.323 protocol. As a result, to identify exactly each protocol message belongs to which session and what is the message type during that session, each 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.

$$\text{Message Type} = M_{ij} \tag{1}$$

During the signalling session, if client 1 sends message to client 2, 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 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}.

As each message is defined by the number of session and the number of the message during the session, therefore all the SIP, RSW, and H.323 sent and received messages behaviours are presented respectively as follow:

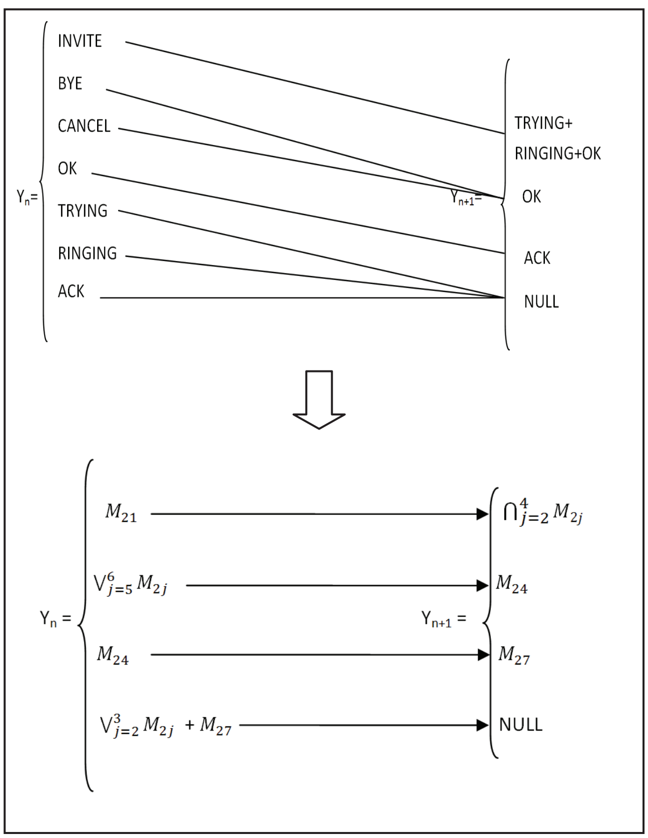


Fig. 1. The behaviour of SIP sent and received signalling messages.

Based on the behaviour of SIP sent and received signalling messages, when client 1 sends INVITE message to client 2, the latest replies by sending TRYING message first, followed by RINGING and OK messages. By representing the messages mathematically, INVITE message in the first SIP signalling message based on the arrangement of the signalling messages, thus INVITE message can be represented as M_{21}. In the same way, each of TRYING, RINGING, and OK messages have the second, third, and fourth order as mentioned before, so each of them can be respectively represented as: M_{22}, M_{23}, and M_{24}. Since all the three messages are acted as a reply for INVITE message, so if Y_n is the message M_{21}, the function Y_{n+1} is $M_{22} \cap M_{23} \cap M_{24}$ which can be written as: $\bigcap_{j=2}^{4} M_{2j}$

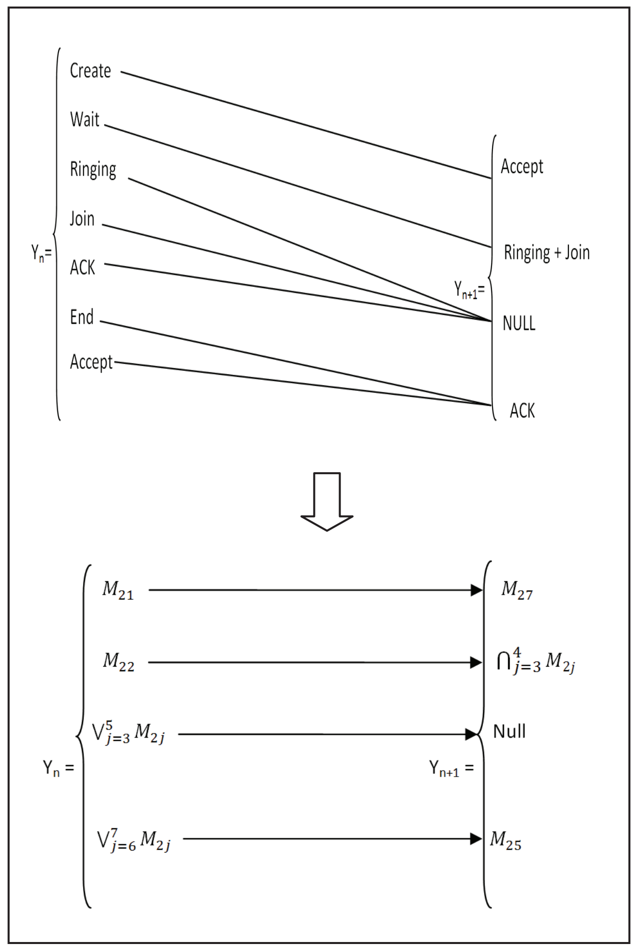


Fig. 2. The behaviour of RSW sent and received signalling messages.

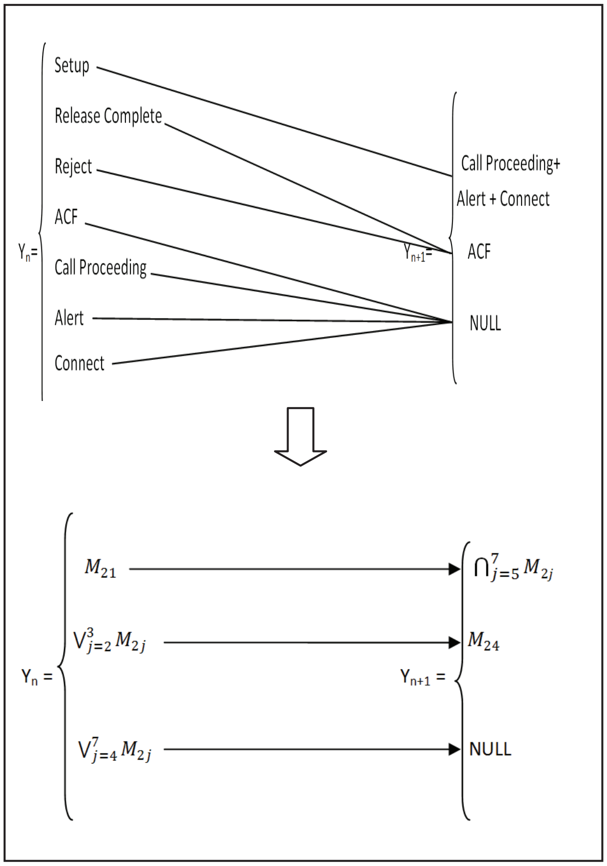


Fig. 3. The behaviour of H.323 sent and received signalling messages.

In media session, j presents the order of a certain media message. J varies from 1 to r where 1 presents the order of the first media message, r is an integer positive number presents the order of the last media message respectively. Assuming that N is the number of messages, so number of 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 each of SIP, RSW, and H.323 clients are carried by RTP header in the media session, by assuming that payload is presented by PL, so

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

4 Session Time Analysis

In media conferencing Environment, two main sessions has to be considered which are signalling session and media session. Signalling session is divided into two sessions; setup session and teardown session. The delay time of each message is the difference between the message sending time from client 1 and the message's receiving time by client 2. 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.

$$T_{M_{2j}} = T_{R\_M_{2j}} - T_{S\_M_{2j}} \quad (4)$$

4.1 Setup Session Time Analysis

In order to calculate the time spent to complete the call setup session, the time difference between the first message sent by client 1 and the last message received by client 2 during the setup session should be measured.

In case of having two SIP clients, the setup session time has been found by calculating the time difference between ACK message arrival time and INVITE message sending time.

$$T_{Setup_{SIP}} = T_{R\_ACK} - T_{S\_INVITE} \quad (5)$$

Hence,

$$T_{Setup_{SIP}} = T_{R\_M_{27}} - T_{S\_M_{21}} \quad (6)$$

While in case of having two RSW clients, the setup session time has been found by calculating the time difference between Join message arrival time and Create message sending time.

$$T_{Setup_{RSW}} = T_{R\_Join} - T_{S\_Create} \quad (7)$$

Hence,

$$T_{Setup_{RSW}} = T_{R\_M_{24}} - T_{S\_M_{21}} \quad (8)$$

In order to find the setup session time for two H.323 clients, the time difference between Connect message arrival time and Setup message sending time has to be calculated.

$$T_{Setup_{H.323}} = T_{R\_Connect} - T_{S\_Setup} \quad (9)$$

Hence,

$$T_{Setup_{H.323}} = T_{R\_M_{27}} - T_{S\_M_{21}} \quad (10)$$

4.2 Teardown Session Time Analysis

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

$$T_{Teardown_{SIP}} = T_{R\_OK} - T_{S\_BYE} \quad (11)$$

Hence,

$$T_{Teardown_{SIP}} = T_{R\_M_{24}} - T_{S\_M_{25}} \quad (12)$$

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

$$T_{Teardown_{RSW}} = T_{R\_ACK} - T_{S\_End} \quad (13)$$

Hence,

$$T_{Teardown_{RSW}} = T_{R\_M_{25}} - T_{S\_M_{26}} \quad (14)$$

In order to calculate the time spent to complete the H.323 call teardown session, the time difference between End message sending time and ACK message arrival time should be calculated.

$$T_{Teardown_{H.323}} = T_{R\_ACF} - T_{S\_Release\ Complete} \quad (15)$$

Hence,

$$T_{Teardown_{H.323}} = T_{R\_M_{24}} - T_{S\_M_{22}} \quad (16)$$

4.3 Media Session Time Analysis

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

$$T_{M_{3j}} = T_{R\_M_{3j}} - T_{S\_M_{3j}} \quad (17)$$

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

$$T_{Media_{SIP}} = T_{S\_BYE} - T_{R\_ACK} \quad (18)$$

$$T_{Media_{SIP}} = T_{S\_M_{25}} - T_{R\_M_{27}} \quad (19)$$

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

$$T_{Media_{RSW}} = T_{S\_End} - T_{R\_Join} \quad (20)$$

$$T_{Media_{RSw}} = T_{S\_M_{26}} - T_{R\_M_{24}} \quad (21)$$

In order to calculate the time spent to complete the H.323 media session, the time difference between Release Complete message sending time and Connect message arrival time (once the call answered) has to be calculated.

$$T_{Media_{H.323}} = T_{S\_Release\ Complete} - T_{R\_Connect} \quad (22)$$

$$T_{Media_{H.323}} = T_{S\_M_{22}} - T_{R\_M_{27}} \quad (23)$$

5 Conclusion

This paper provided a mathematical analysis of the behavior for each of SIP, RSW, and H.323 signaling protocols in terms of the signaling and media messages, and the time of each call setup, call teardown, and media sessions.

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