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A Comparative Study of VoIP, MCS, Instant Messaging Protocols and Multimedia Applications

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***Abstract*— Nowadays, Multimedia Communication has been developed and improved rapidly in order to enable users to communicate between each other over the Internet. In general, the multimedia communication consists of audio, video and instant messages communication. This paper surveys the functions and the privileges of different VoIP protocols (i.e. InterAsterisk eXchange Protocol (IAX), Session Initiation Protocol (SIP), and H.323 protocol), Multimedia Conferencing System (MCS) (i.e. Real Time Switching Control Protocol (RSW) and Multipoint File Transfer System (MFTS), and multimedia applications (i.e. ISO MPEG-4 standards). As well as, this paper make some comparisons among them in terms of codec, transport protocol, call setup format, etc.**

***Index Terms*—Multimedia, VoIP, Instant messages (IM), Multimedia Conferencing Systems (MCS), InterAsterisk eXchange Protocol (IAX), Session Initiation Protocol (SIP), H.323 protocol, Multipoint File Transfer System (MFTS), Real Time Switching Control Criteria (RSW), ISO MPEG-4 standards**

I. INTRODUCTION

Over the last few years, the needs to provide the communication facilities among participants everywhere and every time via computer network systems have been increased. These network systems enable the use of multimedia applications (i.e. ISO MPEG-4 standards) [19] with many kinds of media data, such as audio, video, graphics, images, and text. This rapid expansion and potential underlies the significance of the interworking. Multimedia technology promises to make smooth and very effective interactions among people in different geographical areas [33]. However, the provided multimedia services must be improved.

In recent years, Voice over IP (VoIP) technologies [37] has been developed and many significant progresses have been done in research and commercially. VoIP allows many users to make VoIP phone calls instead of the Public Switched Telephone Network (PSTN) through such technologies as InterAsterisk eXchange Protocol (IAX) [1][5], Session Initiation Protocol (SIP) [12], and H.323 protocol [25][26].

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VoIP can offer a higher quality and yet more reasonable phone service than PSTN. The telecommunication industry is going towards using VoIP as their main phone infrastructure [37]. VoIP services become so popular in the last few years because it is inexpensive compared to the traditional telephony. VoIP can be integrated with other services, such as video conferences, instant messages and presence services.

On the other hand, instant messaging (IM) [29] is a form of online communication that provides a real-time interaction through personal computers or mobile computing devices. Users can transmit and receive messages privately, similar to e-mail, or join group conversations. It has became one of the most common and significant applications of the Internet, causing people to desire to stay connected to the Internet for a long time and allow them to exchange images, audio and video files, and other attachments [30] by using many protocols, such as EXtensible Messaging and Presence Protocol (XMPP) [31].

Multimedia Conferencing System (MCS) [7][8] is a system deals with the digital video, audio, and text data. It transfers these data in real time throughout the network as well as realizing the face-to-face visual meeting by utilizing the fine interactive and management function which are provided by computer system [21]. MCS uses the Real-time SWitching Control Protocol (RSW) [9] to handle a multipoint-to-multipoint multimedia conferencing sessions in terms of audio/video conferencing, whereas the Multipoint File Transfer System (MFTS) [32][34] is used for the same purpose in terms of document conferencing.

Several signaling protocols and techniques are used to help bridging the gap between the endpoints, such as H.323 Protocol, SIP protocol [36], IAX protocol, RSW protocol, MFTS, XMPP protocol, etc. these protocols provides video, audio, data and instant messaging communication among participants [34]. In order to provide and enable the interworking between two or more dissimilar signaling protocols or standards, a translation module must exist in between in order to translate the different control options and instant messages transfer.

This paper is organized into 7 sections; **II, III, IV,** and **V** briefly describe the VoIP protocols, MCS protocols, IM protocol, and ISO MPEG-4 standard respectively. **VI** is a comparison among VoIP, MCS, IM, and multimedia application protocols. And **VII** is a summary of this paper and our planned future research.

II. VoIP PROTOCOLS

A. Session Initiation Protocol (SIP)

SIP is an application-layer control protocol [11] that can establish, modify, and terminate multimedia sessions (conferences) such as Internet telephony calls [14][25][26][27]. SIP can also invite participants to already existing sessions, such as multicast conferences. Media can be added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, which supports personal mobility-users can maintain a single externally visible identifier regardless of their network location [12]. SIP protocol enables Internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share. For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. 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 [23][28].

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) [3][37]. The User Datagram Protocol (UDP) [2] is a transport protocol used to transfer audio and video data [4]. SIP protocol has many features such as the service of text-based which allows easy implementation in object oriented programming languages, flexibility, extensibility, less signaling, transport layer-protocol neutral and parallel search [22][23][24].

B. InterAsterisk eXchange Protocol (IAX)

In (2004) Mark Spencer [5] has created the Inter-Asterisk eXchange (IAX) protocol for asterisk that performs VoIP signaling. Streaming media is managed, controlled and transmitted through the Internet Protocol (IP) networks based on this protocol. Any type of streaming media could be used by this protocol. However, IP voice calls are basically being controlled by IAX protocol [14]. Furthermore, this protocol can be called as a peer to peer (P2P) protocol that performs two types of connections which are Voice over IP (VoIP) connections through the servers and Client-Server communication. IAX is currently changed to IAX2 which is the second version of the IAX protocol. The IAX2 has deprecated the original IAX protocol [5]. Call signaling and multimedia transport functions are supported by the IAX protocol. In the same session and by using IAX, Voice streams (multimedia and signaling) are conveyed. Furthermore, 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 for all the calls between two IAX nodes. Over a single link, IAX provides multiplexing channels [11].

IAX is a simple protocol in such a way Network Address Translation (NAT) traversal complications are avoided by it [8]. The Mini and Full frames are sent between two endpoints A and B. Each audio/video flow is of IAX Mini Frames (M frames) which contains 4 byte header. The flow is supplemented by periodic Full Frames (F Frames) includes synchronization information. UDP transport protocol is used by IAX to transfer audio and video data [4].

C. H.323 Protocol

H.323 is an umbrella standard that provides well-defined system architecture [10], and implementation guidelines that cover call set-up, call control, and the media used in the call [24][25][26]. 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 [27].

III. MULTIMEDIA CONFERENCING SYSTEM PROTOCOLS

A. Real-time Switching (RSW) Control Criteria

Real-time SWitching (RSW) control criteria is a control protocol used to handle a multipoint-to-multipoint multimedia conferencing sessions. RSW control protocol was developed in 1993 as a control mechanism for conferencing by the Network Research Group in school of computer sciences, Universiti Sains Malaysia (USM) [9]. RTP protocol is used by RSW control protocol to carry audio and video data through multimedia conferencing. UDP transport protocol is also used by RSW to transfer audio and video data. The 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 [6][13]. There are several advantages for the RSW control criteria [9] such as Equal Privileges, First Come First Serve, First come first serve with time-out, Organizer Main Site and Restricted Active site.

B. *Multipoint File Transfer System (MFTS)*

The Multipoint File Transfer System (MFTS) is a file distribution system based on the client-server architecture [20][38]. The MFTS server is a distribution engine, which is responsible to handle the document transformation issues, such as file attachment, image sharing, and instant messaging exchange among the various MFTS clients. The Multimedia Conferencing System (MCS) [34] has adopted the MFTS product [35] for the Document Conferencing unit (DC), which is a network component that enables user communications related to file sharing and instant messaging interaction [32].

IV. INSTANT MESSAGING PROTOCOLS

The eXtensible Messaging and Presence Protocol (XMPP) [29] is a standard specified by the IETF for carrying instant message service. It is an open XML protocol for a real-time messaging, presence, and request/response services. First, Jabber open-source community proposed and introduced XMPP and it is still under the development. After that, the Internet Engineering Task Force (IETF) approved and archived it in many Internet specifications. The XMPP architecture consists of three elements, XMPP client, XMPP server and gateways to foreign networks. Transport Control Protocol (TCP) is used by XMPP to transmit and carry media sessions [30]. The developers have been added media session capabilities to XMPP clients which have been defined as an XMPP-specific negotiation protocol called Jingle [JINGLE]. However, Jingle has been designed to easily map to SIP for communication through gateways or other transformation mechanisms [39].

V. ISO MPEG-4 STANDARD: MULTIMEDIA APPLICATION

The recent ISO MPEG-4 standards [15][16][17] target a broad range of low-bit rates multimedia applications: from classical streaming video and TV broadcasting to a very interactive applications with dynamic audio-visual scene customization. In order to reach this objective, advanced coding and formatting tools have been identified in the dissimilar components of the standard ISO 14496; such as audio, visual, and Systems, which can be constructed according to profiles and levels to meet several application needs. A core part of the MPEG-4 multimedia framework is the "Delivery Multimedia Integration Framework" [18]. DMIF provides content location independent methods for creating and controlling MPEG-4 audiovisual sessions and access individual media channels over RTP/UDP/IP.

VI. A COMPARISON OF SIP, H.323, IAX, RSW, XMPP, MFTS, AND DMIF PROTOCOLS.

In this section, we will compare among VoIP, MCS, IM, and multimedia application protocols in terms of call setup format, media transport, transport protocol, codec. Table I shows the comparison among multiple session protocols.

TABLE I.
A COMPARISON OF MEDIA SESSION PROTOCOLS

	Call Setup	Transport Protocol	Media Transport	Codec
SIP	Invite→ ←200Ok Ack→	TCP/UDP	RTP/SRTP	Any IANA-Registered Codec
H.323	Setup→ ←Connect Ack→	TCP/UDP	RTP/SRTP	Any codec
IAX	New→ ←Accept Ack→	UDP	mini frames	G.711, GSM, G.723, etc
RSW	Create conf→ Notify→ ←Join	TCP/UDP	RTP	G.711, GSM, G.723, etc
MFTS	-	TCP	-	-
XMPP	Session-initiate→ ←IQ-result	TCP/UDP	RTP	G.711, Opus, Speex.
DMIF	DS_Session SetupRequest→ ←DS_Session SetupConfirm	UDP	RTP	G.711, G.723.

VII. CONCLUSION

This paper surveys the functions and the privileges of different VoIP protocols (i.e. IAX, SIP, and H.323), MCS protocols (RSW and MFTS), IM protocols (XMPP), and multimedia applications (ISO MPEG-4 standards). In this paper, we made some comparisons of these protocols in terms of codec, transport protocol, call setup format, etc. We can observe that each protocol has its own privileges that differ from the others. In the future, we will do another comparison in terms of quality of services (packet delay, packet loss, jitter, and packet reordering), bandwidth consumption, signaling messages, services, extensibility, scalability, etc.

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