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An Effective Method For Audio Translation between IAX and RSW Protocols

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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 and video communication. However, this paper focuses on audio streams. The audio translation between protocols is a very critical issue due to solving the communication problems between any two protocols, as well as it enables people around the world to talk with each other at anywhere and anytime even they use different protocols. In this paper, a proposed method for an audio translation module between two protocols has been presented. These two protocols are InterAsterisk eXchange Protocol (IAX) and Real Time Switching Control Protocol (RSW), which they are widely used to provide two ways audio transfer feature. The result of this work is to introduce possibility of interworking together.

Keywords**—**Multimedia; VoIP; Interworking; InterAsterisk eXchange Protocol (IAX); Real Time Switching Control Criteria (REW)

I. Introduction

MULTIMEDIA communication has been developed and improved to be the basic and essential service in order to satisfy the needs of the internet users [12, 18, 19, 20]. It has appeared to be more and more applicable in distributed environments [10, 21]. There are various protocols that control and signal the calls of the internet telephone [1]; such protocols existed in Internet Protocol (IP) telecommunication. For data and signaling, two major protocols are considered in the field of the multimedia conferencing such as RSW and IAX protocols. Video functionality, similar quality of service and competitive voice are provided by these two protocols. The two protocols have been widely exploited and utilized into many different methods [15]. This work proposes a translation module between audio streams of IAX protocol and RSW control protocol. Generally, the procedure to a translator between the both protocols is starting from the first network to perform transferring the data into a first protocol. And then the second network to perform transferring the data into a second protocol. Finally, the translation server to maintain translation information for a protocol translation, this translation occurs between the first and the second protocols [15]. IAX and RSW audio transfer protocols have many differences in handling and exchanging voice packets between each other. In addition, different protocols and techniques are used by each of these protocols throughout audio exchange.

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The data size in one packet of IAX protocol (12 bytes) is much smaller than the data size in one packet of RSW control protocol (64 bytes). When the data is attempted to be exchanged among each other by the two protocols, the data could not be understood by each of the two protocols. The reason is due to the different data sizes and different application headers for both protocols. Moreover, different techniques and protocols are used by each protocol so that the audio data could be carried along. The objective of this paper is to overcome the translation problem between the audio streams of the two protocols (IAX and RSW) by proposing an audio translation module, which will be able to act as a translator between IAX and RSW audio streams. This translation module will help bridging the gap between IAX and RSW by providing interworking capability between the two audio streams.

There are two sides accompanied with the translation between the two protocols: the control signaling and audio stream translation. The control signaling translation between two protocols, more specifically between RSW control protocol and IAX protocol has already done [6,7]. And the audio stream translation, which is the proposed work in this paper. The performed audio stream translation is true while this research concentrates on the two protocols (RSW control protocol and IAX protocol).

II. Multimedia Protocols

A. *RSW Control Protocol*

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, University Sciences Malaysia (USM) [9]. The Real-Time Protocol (RTP) protocol [3] is used by RSW control protocol to carry audio and video data through multimedia conferencing. The User Datagram Protocol (UDP) transport protocol [2] 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. The RSW audio packet format consists of four parts, which are the IP header [2], the UDP header, the RTP header and the RSW payload. Audio packet format of the RSW protocol is shown in Fig.1.

IP (20 bytes)	UDP (8 bytes)	RTP (12 bytes)	RSW media frame (64 ytes)

Fig. 1 RSW Audio Packet Format

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]. The IAX audio packet format consists of four parts, which are the IP header, the UDP header, the IAX header [5] and the IAX payload [5]. Audio packet format of the IAX protocol is shown in Fig. 2.

IP (20 bytes)	UDP (8 bytes)	IAX header (4 bytes)	IAX payload (12 ytes)

Fig. 2 IAX Audio Packet Format

C. Media Transfer in IAX/RSW Protocol

In order to transmit the packets from the sender to the receiver, an audio codec should be used. In this paper, GSM codec is proposed in order to convert the analog audio to digital form [17]. Then, the digital audio will be compressed to decrease the consumption of network bandwidth needed to pass on the speech to the receiver. GSM audio codec has the bandwidth (13.2 kb/s). However, while sending the codec data across an IP network, it is going to increase the used bandwidth. While dealing with voice, it is better not to introduce too much latency. For example, sending voice frames every 20 ms, which means every 50 frames have to be sent in one second. Dividing 13.2 kb/s by 50 will give 33 bytes. As a result, 33 bytes of voice data will be sent per one frame. In addition, IP header (20 bytes a packet), UDP header (8 bytes a packet) adding the RTP protocol header (12 bytes a packet). This codec will be used not only for the IAX protocol, but also for the RSW control protocol. In this paper, the two audio streams protocols will use the same codec to transmit and receive the audio packets between the clients.

III. Proposed Solution for the Audio Translation

In general, some of the VoIP protocols such as IAX use small chunks of data (12 bytes) in its frame when it transmits audio over IP, while others such as RSW use big chunks of data (64 bytes). This issue will cause a problem when trying to internetwork between protocols. The idea of a translation module between audio streams of protocols that use different sizes in data will be discussed in this paper. Even though, the proposed method should work between any two protocols, but this paper concentrates on audio translation between IAX and RSW protocols. These two protocols use different techniques and different protocols in audio exchange. As well as, they use different mechanism in handling voice packets. Fig.3 shows the translation in IAX-RSW.

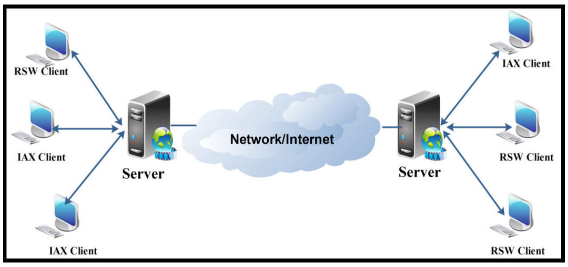

Fig. 3 IAX-RSW Translation

The translation module has two ways; the first way is the translation from RSW audio stream to IAX audio stream. The second way is the translation from IAX audio stream to RSW audio stream. For each of the two ways, the packets will be prepared in two buffers (IAXtoRSW and RSWtoIAX buffers) which are supported by the UDP socket [16]. The UDP socket makes use of the port numbers (source port number and destination port number) [16]. UDP sockets also provide two buffers for both source and destination. The buffer size is 4096 bytes. The audio translation procedures from IAX to RSW are multiplexing, removing IAX header, and adding RTP header. While, from RSW to IAX are partitioning, removing RTP header, and adding IAX header. The relationship between IAX and RSW packets is shown in Fig.4 which describes the translation processes for the two ways. Firstly, Change the packet format of IAX audio stream to the packet format of RSW audio stream. Secondly, Change the packet format of RSW audio stream to the packet format of IAX audio stream.

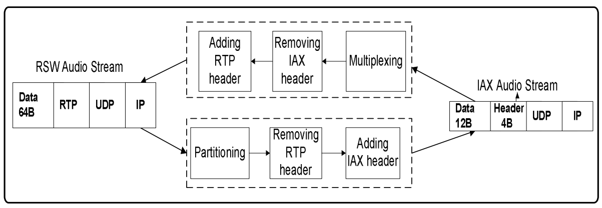

Fig. 4 Translation Procedures

IV. DISCUSSIONS

This section will explain the two audio streams translation methods, which are RSW to IAX and IAX to RSW. In fact, there are two processing steps in the audio translation method from IAX to RSW/RSW to IAX. The first step is collecting the IAX/RSW audio packets; the packets are collected in the IAXtoRSW/RSWtoIAX buffer which is supported by UDP socket. This buffer should be in the client. The second step is the translation processes. This translation processes should be done in the gateway.

A. Multiplexing and Partitioning Processes

The multiplexing process (IAX to RSW) occurs two times. The first multiplexing will be inside the first gateway. Data coding will be done by using GSM codec while sending these data. The GSM codec packets have the data size of 33 bytes [17]. In this case, the IAX packets (12 bytes) should be multiplexed to the data size of 33 bytes. The second multiplexing occurs when the packets are received by the second gateway; the packets will be at the size of 33 bytes. But the RSW packets have the size of 64 bytes. So, the GSM codec packets (33 bytes) should be multiplexed again to the size of 64 bytes. Furthermore, the partitioning process (RSW to IAX) happens also twice. The first partitioning should occur in the first gateway. The RSW packets (64 bytes) should be divided to the data size of 33 bytes. The second partitioning occurs when the packets are received by the second gateway; the packets will be at the size of 33 bytes. But the IAX packets have the size of 12 bytes. So, the GSM codec packets (33) bytes should be divided again to the size of 12 bytes.

B. Adding and Removing IAX]RTP header Processes

Adding RTP header process is one of the translation processes. It occurs in the gateway when sending audio packets from IAX client to RSW client. IAX protocol does not use RTP protocol for carrying audio packets during transmission whereas RSW uses it. So, RTP header should be added when translation supposed to be done from IAX to RSW. But, in the translation from RSW to IAX, we notice that IAX protocol only uses UDP protocol as a transport protocol. So, RTP header should be removed. In addition**,** Adding IAX header process occurs in the gateway when sending audio packets from RSW client to IAX client. The IAX header should be added to the audio packets to give significant information as the source call number and timestamp. But, when sending audio packets from IAX client to RSW client, the IAX header should be removed. Fig.5 describes the architecture of the translation module.

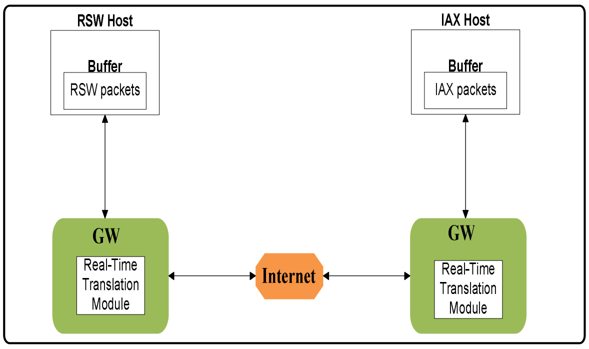

Fig. 5 Translation Module Architecture

V. SUMMARIES

For guaranteeing a seamless end to end connectivity for IAX-RSW translation, we have proposed solution to these interworking problems by doing an audio translation module. Using the proposed Translation Module will enable VoIP protocols uses different sizes of data chunks to exchange audio streams. The translation module can be a base for any two different protocols or more. The proposed method came as a translation audio streams module that can be used when provide IAX-RSW full interworking. It will be a valuable research if the translation module that depends on media transfer part (which is this performed work) combined with translation module that depends on the signaling part. So, we will get a completed result. Currently, the researchers are implementing the prototype system of the translation module architecture of IAX and RSW. The next step of this work is to evaluate and analyze the prototype system and obtained knowledge will show that relation among media, resource and session parameters. However, the required knowledge for translation entity for IAX and RSW server is acquired.

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