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A Comparative Study between Inter-Asterisk Exchange and Jingle Protocols

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Abstract

In the last few years, multimedia communications has been developed and improved rapidly in order to enable users to communicate between each other over the internet. In general, the multimedia techniques consist of instant messages, audio, and video chatting services. However, users cannot phonetically communicate with each other unless they use the same chatting applications since each chatting application has its own control protocol to handle the call setup, the real time media transmission, and the call teardown sessions. The only way to enable the users to communicate phonetically using different chatting applications is to design a new architecture by proposing an interworking module between the control protocols used by the applications. The interworking module between at least two different protocols is a very critical issue as it solves the data transmission problems when using different chatting applications to create a voice call, and demonstrates that the voice conferencing between heterogeneous control protocols identifies the interest of combining the features and taking the advantages of more than one protocol in a single network. The two chosen protocols in this paper are: Inter-Asterisk exchange Protocol and the extension of extensible Messaging and Presence Protocol. Each protocol differs from the other one in many terms. Thus, this paper clarified the main differences between IAX Protocol and Jingle Protocol by doing a comparison in terms of registration, URI format, signals, transport methods, audio and control packets, bandwidth, scalability, functionality, services, etc. This comparison is due to define the common and the unique features to ease proposing the IAX-Jingle mapping architecture, signaling and media transfer modules. So, people around the world will be enabled to talk with each other without caring whether they are IAX or Jingle users.

Keywords: IAX, Jingle, Multimedia, Signaling Protocols, VoIP

1. Introduction

With the appearance of numerous multimedia conferencing and Voice over Internet Protocols[1–9], 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. This paper defines the attributes of IAX and Jingle protocols because of their services compared with the other signaling protocols. Therefore, the objective of this paper is mainly to make a comparative study between IAX and Jingle protocols in order to determine the common and unique features in terms of audio services. This paper does not cover video conferencing and document conferencing services (instant messaging, file attachment and image sharing) since IAX is a VoIP Protocol, despite it can be used for any type of streaming media, but it is mainly designed for IP voice calls.

In this section, we will briefly define IAX and Jingle protocols with regard to the main functions and other properties that indicate their preference compared with the rest of the protocols.

1.1 IAX Protocol

Mark Spencer has created the Inter-Asterisk exchange (IAX) protocol for asterisk that performs VoIP signaling[10]. IAX is supported by a few other soft-switches,

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(Asterisk Private Branch exchange) PBX systems[11], and soft-phones[12]. Any type of streaming media can be managed, controlled and transmitted through the Internet Protocol (IP) networks based on IAX protocol. However, IP voice calls are basically being controlled by IAX protocol. Currently, IAX has been changed to IAX2 which is the second version of the IAX protocol[13,14].

IAX2 has deprecated the original IAX protocol[15]. Call signaling and multimedia transport functions are supported by the IAX protocol. In the same session and by using IAX, Voice streams (media flow and signaling) are conveyed in binary format. 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 by two IAX nodes for the whole calls. Over a single link, IAX provides multiplexing channels[16, 17].

IAX protocol is used for many purposes[18]. Firstly, it is to minimize bandwidth usage for both control and media transmissions with specific emphasis on individual voice calls, secondly, to provide Network Address Translation (NAT) transparency, thirdly, to support the ability to transmit dial plan information, and lastly, to support efficient implementation of intercom and paging features.

Asterisk is the best known VoIP related Free Software server which is mostly used as a VoIP solution[19]. The easiest way to try out Asterisk's traditional telecom features is by using a computer equipped with a modem (supported by Linux) or an Integrated Services Digital Network (ISDN) adapter. Asterisk supports many protocols, such SIP, IAX, and Skinny Call Control Protocol (SCCP). Asterisk uses a concept of channels where different terminals can connect with different VoIP protocols. Asterisk server can also be used as a media gateway or an application server[20, 21].

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[22,23], H.323[24], MGCP[25] and RSW[26,27] in terms of quality of services.

Apart from its simplicity, the advantages of the IAX protocol over other VoIP protocols[28,29] are as follows:

- IAX mixes the signaling and media paths. The separation of media and signaling is possible once the connection has been successfully established.
- IAX does not depend on other protocols for media transmission, IAX handles media streams itself. Various media types may also be sent by IAX: Voice, video, image text, and HTML.
- IAX has defined reliable and unreliable messages. IAX's unreliable messages are media sessions which are not acknowledged nor retransmitted if lost in the network. IAX reliability is ensured for signaling messages because of its several IAX application identifiers maintained by the IAX clients during signaling session. These messages should be acknowledged; if not, these messages are retransmitted.
- NAT traversal is not a big issue with IAX. IAX's signaling messages never carry IP addresses.
- IAX measures the network performance and this measured information can be shared with other IAX clients during an active session.
- IAX client will be notified about its peers.
- IP security modules can be deployed together with IAX. IAX entertains exchange of shared keys.
- IAX authentication is implemented because of its exchange of authentication requests, which carries a security challenge. This authentication challenge should be replied by the remote IAX client and encrypted according to the adopted encryption method during that particular session. If encryption negotiation has failed, the call should be terminated.
- IAX has a trunking property, so several communications can be multiplexed into this data stream.

1.2 Jingle Protocol

The extensible Messaging and Presence Protocol (XMPP)[30] is a standard specified by the IETF for carrying instant message service. XMPP is an open extensible Markup Language (XML) protocol for a real-time messaging, presence, and request/response services, and it is an out-of-band signaling protocol. 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[31]. The XMPP architecture consists of three elements, XMPP client, XMPP server and gateways to foreign networks. The developers have added media session capabilities (which have been defined as an XMPP-specific negotiation protocol called Jingle) to XMPP clients[32–34].

Jingle has been designed to support many types of applications, such as voice and video conferencing, file transfer, application sharing, and others. The security requirements have to be considered before starting the transferring of data between two nodes. However, Jingle has been designed to easily map or plan to SIP for communication through gateways or other transformation mechanisms so that the millions of deployed XMPP clients can be added onto existing VoIP networks, rather than limiting XMPP users to a separate and distinct network.

The XMPP architecture is composed of three elements, XMPP client, XMPP server and gateways to foreign networks[35]. The main responsibilities of the server are in managing the connections to and from sessions for authorized clients, servers, and other entities, and also routing appropriately-addressed XML stanzas among such entities over XML streams.

Most clients connect directly to a server over a TCP connection and use XMPP to take full advantage of the functionality provided by a server and by any associated services. A gateway is a special-purpose server-side service; its primary function is to translate XMPP into the protocol used by a foreign (non-XMPP) messaging system, as well as to translate the return data back into XMPP.

Jingle protocol offers many properties via Jabber Company, such as service discovery, entity capabilities, data forms, multi user chat, file transfer, HTTP binding, publish subscribe, geo location, reliable message delivery, and many other features.

Just as IAX protocol has many benefits, Jingle also offers many privileges in terms of provided services, variety of codec's, NAT traversal, end to end encryption. Such properties are:

- Jingle is supported by many clients, such as Asterisk PBX, FreeSwitch, Pidgin, Google Talk for Gmail[36], Miranda IM, Coccinella, Jitsi, and others.
- The unique similar project is Skype which is a closed and non inter-operable network. Skype Network works in a similar way except that on Skype you cannot choose whether share your bandwidth or not. As a closed protocol, you cannot do anything about it. On Jingle you can choose to share with: ◦Everyone, ◦nobody, ◦only buddies, only white-list, black-list, etc.
- Jingle Relay Nodes uses the benefit of the embedded security and flexibility of XMPP to provide the same functionality of Relay, but in a much simpler process and minimal implementation, so we can finally have XMPP Clients doing VoIP with a pure XMPP Stack Client.
- Jingle nodes promote the beauty of P2P, but without dropping the Jingle Responsibility with the end user, which is call completion.
- Jingle nodes remove the limitation of the amount of streams that can be decoded by the client.
- Jingle nodes remove the limitation of the client's downstream and upstream bandwidth.
- Jingle nodes provide scalable text, audio, and video codec's.
- Jingle has the ability to cross any NAT device, NAT Traversal is provided for users that do not have STUN/TURN support, and also for users with STUN/TURN support that the negotiation failed.
- The MUC server does not need to provide any infrastructure for Multi-User Jingle.
- Jingle provides end-to-end encryption in the whole sessions.

2. A Comparison between IAX and JINGLE Protocols

IAX and Jingle protocols have many differences which will be mentioned in details later. Such differences are control (call setup and call teardown) messages, audio packet for real time voice transmission, header, URI format, transport protocols, media transports, codec's, modularity, types of connections, registration procedures, scalability, bandwidth, descriptors, multiparty calls supporting, and others.

Due to the differences between IAX and Jingle protocols, this paper will attempt to overcome the heterogeneity problems faced by interwork IAX with JINGLE. This section describes the terms of differences represented by signaling messages, media transfer, transport methods, codec's, bandwidth and others.

2.1 Registration

The registration/authentication is optional in IAX. In order to make calls between IAX clients, the caller endpoint should know the IP address of the called endpoint. IP address can be manually provided to the calling endpoint. IAX also provides a peer to peer registration facility to help IAX clients reach the registrant. IAX can use the NEW message type to select authentication type, such as Message Digest Algorithm 5 (MD5), RSA (Rivest, Shamir, and Adleman) algorithm, and Triple Data Encryption

Standard (3DES). For encryption mechanisms, IAX uses either plain text or Advanced Encryption Standard (AES).

The server contains user specific information, such as log in ID, password, URI, online status, and other personal information. The following messages are exchanged between IAX client and its respective server:

- An IAX REGREQ is sent by an IAX client to the first service contact point. An IP address or an FQDN (Fully Qualified Domain Name) is provided to IAX client prior to the registration process.
- The REGREQ message is then received by the server. As a result, the server understands that this message is to authenticate the IAX client, and sends REGISTER message to the IAX client.
- Upon receipt of the REGISTER message, IAX client sends ACK message to the server.
- Once the server receives ACK message, a REGACK is built and sent to the IAX client.
- As a response, IAX client sends an ACK message to the server. At this stage; IAX client is registered with the server and may place and receive calls from remote user client.

Authentication matter is different from Jingle protocol. This is because some steps must be successfully done to start the communication between two endpoints. As mentioned before, the negotiation in Jingle protocol takes place over XMPP protocol which uses TCP as a transport method. For end to end encryption, signaling sessions use Transport Layer Security (TLS) protocol, whereas it is recommended for the media session to use the Secure Real-time Transport Protocol (SRTP).

Hence, to start in the authentication matters, the following messages are exchanged between the server and Jingle to authenticate Jingle client:

- The client establishes the TCP connection to the server and initiates the XML streams.
- The server sends a STARTTLS (Transport Layer Security) extension to the client, including the supported authentication mechanisms.
- The client responds to the STARTTLS command.
- The server informs the client that it is ok to proceed.
- The client and the server complete the TLS setup, and then the client initiates a new stream to the server,
- The server responds by sending a stream header to client along with any available stream features (Simple Authentication and Security Layer protocol (SASL)).
- The client picks up an appropriate authentication mechanism.
- The server sends a Base64-encoded challenge to the client.
- The client responds to the challenge with the credential.
- The server sends another challenge to the client, as the session token.
- The client again responds to the challenge.
- The server informs the client of successful authentication.
- The client initiates a new stream to the server for the application-purpose communication.

2.2 Uniform Resource Identifier

Although both protocols define the name address and the specifications such as server domain name, port, and other contexts, but each protocol presents this information in its own URI in different way and format.

URI address = [name address | address specifications]
Name address = username
Address specification = host name + port + number
Host name = server domain name
IAX URI format is:
iax: [<username>@]<host> [: <port>] [/<number> [?<context>]]
Jingle URI format is:
xmpp: [<user>]@<host> [: <port>] /[<resource>] [?<query>]

2.3 Connection Architecture

IAX can be called as peer-to peer protocol that performs two types of connections. They are VoIP connections through asterisk servers, or it can be called as PBX, and client-server communication. Like IAX, Jingle enables one-to-one, peer-to-peer media sessions between XMPP entities. Jingle is usually executed via client-server architecture, where the client uses the XMPP protocol to access the XMPP server, also the XMPP servers can communicate with each other over TCP Connections[37]. In case of multipoint call where three or more endpoints communicate in real-time over an IP connection, IAX does not specify a certain procedure for enabling multiparty conferencing, while in Jingle it is well defined.

2.4 Header Format

IAX protocol uses full and mini frames which have the size of (12–4 Bytes) respectively, and is adjustable to suit a wide variety of control signals and audio packets. However, Jingle does not have a formal header as the header is variable in length and it depends on the message type and the required information.

IAX protocol has two types of frames; the full frame and mini frame. A full frame sends signals, audio, or video information reliably. This type of frame is the only type that is transmitted reliably. This means that the recipient host has to return a message back to the sending host instantly upon reception. Sometimes, the protocol may require a particular message that can be sent back instantly, otherwise, the recipient must send a clear acknowledgement. After the sending host sends a NEW message, the recipient host must return an ACCEPT message immediately. In this case, it does not require ACK message. Later, when a RINGING message is sent to the caller, Host A must send back an explicit ACK message since the IAX protocol does not require any other message to be returned at that time.

The second type of IAX frames is the mini frame. The mini frame sends media with a minimal protocol transparency. Both full and mini frames have different usages each. The full frame is used during the signaling part of the session, which includes the registration, call setup, call teardown, etc, whereas, the mini frame is used in the media session while transferring the real time media packets between the endpoints. Sometimes, the audio flow is supplemented by periodic full frames that includes synchronization information.

On the other hand, Jingle header is a variable in length per header/ per message. Jingle header may consist of three main fields. The first field is the packet type; this field allows the entities to understand that the packet is a control packet. The second field is the message type; this field defines the function of the control message (i.e., session-initiate, session-accept, etc). The last field is the message information; this field provides some required information for the message type, such as user ID, session ID, etc.

2.5 Signaling Messages

Each of the two protocols has its own signals; these signals differ in IAX from Jingle. However, both protocols have the same goal of using the signals which initiate and terminate the call between two clients. Table 1 shows a comparison between the common call setup and teardown signals of IAX and Jingle protocols.

IAX uses several signals (i.e., NEW, RINGING, ANSWER, HANGUP, etc.,) in order to register and setup the call before starting the audio conferencing between two or more clients, as well as, the signals are used again at the end of the audio transmission to hang up and teardown the call[38].

Table 1. IAX and Jingle call setup and teardown messages

IAX	Jingle
NEW	Session-initiate
ACCEPT	Session-info (ping)
RINGING	Session-info (ringing)
ACK	IQ-Result (ack)
ANSWER	Session-accept
HANGUP	Session-terminate <success/>
BUSY	Session-terminate <busy/>
REJECT	Session-terminate <decline/>

However, there are three state machines provided by Jingle protocol for overall session management, which are the pending, active, and ended states. The pending state occurs between session-initiate and session-accept, while the active state happens after the session-accept and before the session-terminate. The last state machine is the ended state which is done at the end of the session-terminate[39].

Both pending and active states include many actions related to session management, such as the initiating and the terminating sessions, adding and modifying some contents, replacing the transport method, etc.

2.6 Two Endpoints Call Mechanism

IAX starts the call, Endpoint A sends NEW packet to the Endpoint B to place a call, and then wait until it receives the ACCEPT packet from Endpoints B. After the ACCEPT reply, Endpoint A sends ACK packet to Endpoint B to acknowledge that it has received 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 the ACCEPT message. Then, Endpoint B sends ANSWER packet to Endpoint A in order to start the call, and then wait till it sends 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 replies back by sending the Acknowledgment packet (ACK). Figure 1 shows the three main procedures used for the audio conferencing between two IAX endpoints, which are call setup, audio transmission, and call teardown with the steps of each procedure[40].

In Jingle, there are three types of sessions that are used to do the negotiations between two entities. They are respectively session-initiation, session-accept, and session-terminate. To start the negotiations, two entities are involved, the initiator and the responder. In order to setup the call, the initiator sends a session initiation offer to the responder. After the responder has acknowledged receipt of the session-initiate message, it prompts the responder to choose whether he wants to proceed with the session or not. If he wants to proceed, he will select the appropriate interface element and his client will send a session-accept message to the initiator. The initiating client will acknowledge the receipt of the session-accept message.

The aforementioned steps to initiate the call take place over XMPP signaling protocol. After that, both the initiator and the responder can exchange the audio data over RTP protocol. Eventually, one of the clients will terminate the session. The other client will acknowledge the receipt of the session-terminate message and the session is ended. Similar to the call setup, the call teardown takes place over XMPP signaling protocol[41]. Figure 2 clarifies the communication between two XMPP clients.

2.7 Media Transfer

Each of the two protocols has a certain way to support the exchange of the audio packets. IAX protocol uses the mini headers (4 Bytes) to carry the voice packets during the media transferring session. On the other hand, Jingle does not have its certain frames, therefore, RTP header (12 Bytes)[42,43] is used to carry the data during the voice chatting session.

2.8 Codec's Support

The quality and performance of any protocol depends on the use of the codec. Both protocols use numerous codec's[44,45]. The supported codec's is identified in Table 2.

2.9 Transport Methods and Port Numbers

In addition to the aforementioned differences in both protocols, IAX differs from Jingle in the transport protocols

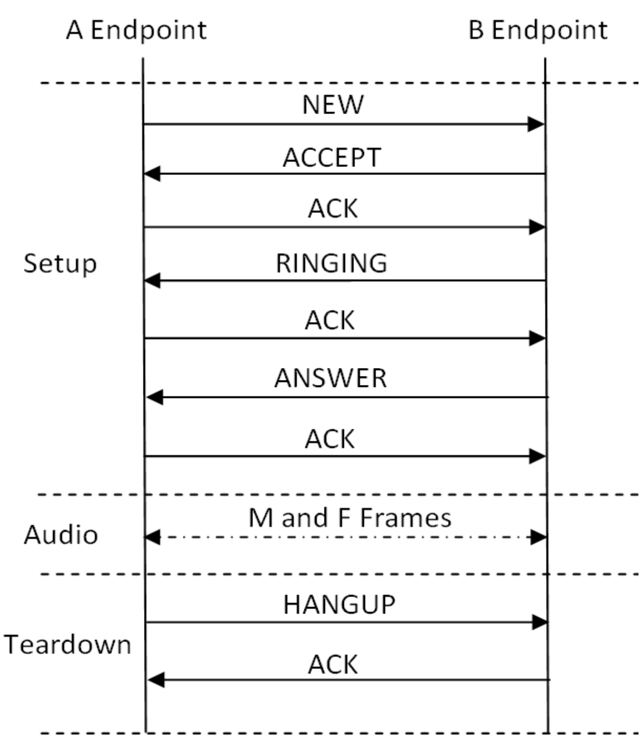

Figure 1. IAX-to-IAX communications[40].

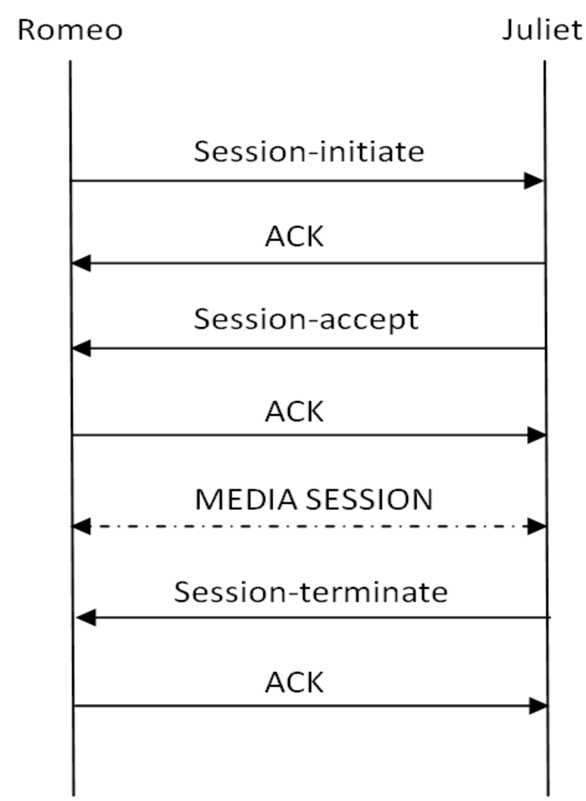

Figure 2. XMPP-to-XMPP communications[32].

Table 2. IAX and Jingle supported codec's

	IAX	Jingle
G.711	√	√
G.723.1	√	×
G.729	√	√
Speex	√	√
GSM	√	×
PCMU	×	×
OPUS	×	√
L16	×	√
ILBC	√	×

and port number used for connecting the client and the server. IAX protocol makes use of only one transport protocol which is UDP. This unique protocol is utilized during the whole three sessions (call setup session, media exchange session, and call teardown session). The same UDP port is used throughout media transmission and signaling information sessions since IAX2 recommends only the UDP port (4569) whereby port (5036) is used by IAX1.

While, Jingle uses two transport protocols, TCP protocol can be used in both signaling and media sessions, whereas UDP is used only during the voice chat (media) session[46]. The Jingle assigns the port (5269) to be used for connections between two servers and the port (5222) to be utilized for client.

2.10 Audio and Control Packets Formats

Both IAX and Jingle protocols have two types of packets; control packet and audio packet. The control packet differs from the audio packet in form and function and each one of them is used within certain sessions.

The IAX audio packet format during the media transporting session can be divided into four parts. They are IP header (20 Bytes), UDP header (8 Bytes), IAX mini header (4 Bytes), and the payload (the size of the payload relies on the type of the codec's that the protocol uses). IAX control packet format that is used in the signaling part differs from that of the audio packet. The IAX uses the full header for the signaling messages that includes the registration, call setup, call teardown and other signals, and uses mini header for the real time voice chat.

IAX full header (12 Bytes) contains some required information (i.e., source and destination call numbers, frame type, message type, etc.,), while IAX full header data (12 Bytes) has the control message. Both IAX audio packet and control packet formats with the size of each field of them are shown in Figures 3 and 4.

The Jingle audio packet format during the voice chat session, while transferring the audio between two clients, is comprised of four parts which are IP header (20 Bytes), UDP header (8 Bytes), RTP header (12 Bytes), and the payload (the size of the payload relies on the type of the codec's that the protocol uses). The Jingle control packet that is used in the signaling part differs from that of the audio packet since Jingle uses XMPP protocol during the negotiating sessions which in turn uses TCP protocol to transport the data, whereas RTP is used instead of XMPP to carry the audio packets.

For the formation of the control packet, four fields are required; such fields are IP header (20 Bytes), TCP header (20 Bytes), Jingle header (variable per message type), and the message body which includes the body of the control message. The Jingle message body is also variable as it depends on the message size and type. Both audio and control packets formats of Jingle with its size of each field of each are shown in Figures 5 and 6.

2.11 Network Address Translation

In order to avoid NAT problems, IAX or IAX2 uses UDP transport protocol, normally on port 4569, (IAX1 used port 5036), and both signaling information and data are sent by using the same protocol. Therefore, IAX has less

IP (20 Bytes)	**UDP (8 Bytes)**	**IAX Mini Header (4 Bytes)**	**Payload**

Figure 3. IAX audio packet format.

IP (20 Bytes)	**UDP (8 Bytes)**	**IAX Full Header (12 Bytes)**	**IAX Full Header Data (12 Bytes)**

Figure 4. IAX control packet format.

IP (20 Bytes)	**UDP (8 Bytes)**	**RTP (12 Bytes)**	**Payload**

Figure 5. Jingle audio packet format.

IP (20 Bytes)	**TCP (20 Bytes)**	**Jingle Header**	**Jingle Message Body**

Figure 6. Jingle control packet format.

NAT problems and it can pass through routers and firewalls in a better way and offer NAT transparency. Besides, NAT traversal is assisted by using Interactive Connectivity Establishment (ICE) in the Jingle protocol with the purpose of providing robust NAT traversal for media traffic[47].

2.12 Bandwidth

The bandwidth of the first call and the additional calls are calculated for both protocols. In order to compute the required bandwidth for the number of conversations (n) full duplex[48], the below formula should be applied.

$BW_n = BW*n*2$

In order to find the bandwidth of the first call for IAX protocol, the supported codec should be identified with its constraints. In this paper, the recommended codec for both IAX and Jingle to be used is G.711. In general, the payload size of G.711 is 160 Bytes, and 50 packets can be transmitted per second. Firstly, the total size of the packet (TPS) in the first call must be known by determining five parameters; the size of the protocol header in the data link layer L2 (Ethernet), the size of the protocol header in the network layer (IP), the size of the transport protocol header (UDP), the size of the session protocol header (IAX), and the size of the voice payload (VPS).

TPS = Ethernet + IP + UDP + IAX mini header + VPS

TPS = 18 Bytes + 20 Bytes + 8 Bytes + 4 Bytes + 160 Bytes = 210 Bytes = 1680 Bits.

After calculating the total size of packet, the result of TPS is multiplied by the number of packets per second (PPS) in order to find the bandwidth of the first call.

BW = 1680 Bits* 50 = 84 Kbps for the first call.

IAX protocol has a multiplexing feature, so for each additional call, the headers of data link, network, and transport protocol are reused, as a result:

TPS = 4 Bytes + 160 Bytes = 164 Bytes = 1312 Bits.

BW = TPS* PPS = 1312 Bits* 50 = 65.6 Kbps for additional call.

For example, the bandwidth of the voice chat which includes 30 calls is (the bandwidth of the first call + the bandwidth of the additional 29 calls)* 2.

$BW30_{calls}$= (84 Kbps + 65.6 kbps* 29)* 2 = 3972.8 Kbps (Approximately 4 Mbps).

In the same way, the same codec is used to find thebandwidth of the first and additional calls for the Jingle protocol. Unlike IAX, Jingle does not have its own session header; in place of the protocol header, RTP protocol is used to carry the audio packet during the media session, thus:

TPS = Ethernet + IP + UDP + RTP + VPS

TPS = 18 Bytes + 20 Bytes + 8 Bytes + 12 Bytes + 160 Bytes = 218 Bytes = 1744 Bits.

BW = TPS* PPS = 1744 Bits* 50 = 87.2 Kbps for one call.

The bandwidth for 30 calls = (the bandwidth for one call* 30)* 2

$BW30_{calls}$= (87.2 Kbps* 30)* 2 = 5232 Kbps (Approximately 5.2 Mbps).

2.13 Functionality

The differences of functionality terms in both protocols, as defined in Table 3, lead to many issues that need to be processed and solved. Such terms are modularity (issue: Synchronization of the mapping module during signaling and media sessions), request and response format (issue: translation of syntax), address schema (issue: translation needed to match the addressing format), floor control (issue: Admission control need to be addressed), and capability exchange (issue: The control gateway has to deliver common codec for audio and video to the participant).

2.14 Complexity

There are many factors that indicate whether IAX and Jingle protocols are complex or simple. These factors vary between the number of request and response messages, format of encoding, and number of transport protocol used and others. Table 4 shows the comparison between the two protocols in terms of complexity factors.

2.15 Extensibility

IAX open nature permits new payload type and field additions needed for support. Currently, many new information elements and frame types are introduced and can be found in the asterisk source code which is freely available online. However, at variance jingle which has the ability to change the audio and video codec's during the conference; IAX uses a point-to-point codec negotiation mechanism that limits extensibility because every IAX node in a call path must support every used codec to some degree. On the other side, Jingle also has introduced several new elements, particularly, with respect to the extension of RTP header during media session, for instance, the new element <rtp-hdrext>[49].

Table 3. The variation of IAX and Jingle in functionality terms

Functionality	IAX	JINGLE
Modularity	All in one module; combined signaling and media	Separate modules for different functions independent call signaling and media channel
Request and Response Format	Binary Format	Pure XML
Address Schema	SIP-like URI Format	URI Format
Floor Control	No floor control	Support many mechanisms to allow the users to share resources
Capability Exchange	Media description is carried in the format of full frames	Available media handshaking mechanism <Jingle/> or JID)

Table 4. Comparisons between IAX and jingle complexities

Number of Signals	Both protocols have comparable amount of signals vary between 30 and 40 control messages
Format of Encoding	Both Jingle and IAX use binary encoding; this makes implementations substantially simpler and more robust to buffer overrun attacks.
Number of Transport Methods	IAX uses only UDP in all sessions, while Jingle uses UDP during the media session, also TCP can be used during both signaling and media transfer sessions.

2.16 Scalability

IAX is carried on UDP for the whole sessions; hence, no connection state is required. This means that the servers can be based on UDP and operate in a stateless fashion, whereby reducing significantly the memory requirements and improving scalability. Besides, IAX can handle the call loops by using TTL (Time-To-Live) property. While, Jingle connections are TCP based, which means that a server must hold its TCP connections for the entire duration of a call. This can pose serious scalability problems for large servers.

Nevertheless, JINGLE is a general-purpose SIP which uses a loop detection algorithm similar to the one used in Border Gateway Protocol (BGP) which can be performed in a stateless manner[50]. Moreover, both protocols support multiparty conferences with multicast data distribution, and define the procedures of user location via IP address.

2.17 Services

Jingle protocol has a lot of different service discovery features, and also support many scenarios, such as video chatting, music streaming, port allocation tuning, supporting a different signaling protocol. Voice calls from such clients can be gateway to other VoIP services, mobile and landline. GTalk2VoIP is an example of such a service. XMPP/JINGLE/VoIP combination of open standards gives a clear platform for solutions to compete against Skype51, the (proprietary) market-leading software phone system. Many services and attributes are also supported by IAX protocol, for example, World class network infrastructure, easy to use web-based account and service management, voice mail services with Directory, Call Conferencing, IVR (Interactive Voice Response) and Call Queuing, trunking, jitterbuffer, and general NAT issues are supported. Recently, 2talk has launched an IAX2 Beta service aimed for customers with an 'Asterisk' based IP-PBX system including Asterisk 1.2/1.4/1.6, Trixbox, FreePBX, Callweaver, Freeswitch, etc[52].

3. Conclusion

This paper provides a brief explanation about IAX and Jingle protocols by identifying the functions and the privileges of both protocols. The reason for the choice of IAX and Jingle protocols to be compared has been justified. In addition, the paper discusses the main differences between the two protocols by doing a comparison in terms of signals, headers format, audio and control packets, transport methods, bandwidth, scalability, complexity, functionality, services, etc. For future work, it is proposed that both protocols (IAX and Jingle) should be more fully evaluated and quantitative performance metrics to be examined so as to characterize these differences.

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