REAL TIME CONFERENCE GATEWAY FOR HETEROGENEOUS CLIENTS: REAL TIME SWITCHING CLIENTS AND INTER-ASTERISK EXCHANGE CLIENTS

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by

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LIST OF ABBREVIATIONS

IP	Internet Protocol
VoIP	Voice over Internet Protocol
PSTN	Public Switched Telephone Networks
RSW	Real Time Switching
MCS	Multimedia Conferencing System
IAX	InterAsterisk eXchange
UDP	User Datagram Protocol
RTP	Real Time Protocol
NAT	Network Address Translation
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
LAN	Local Area Network
MLIC	Multi-Lan IP converter
DC	Data Compression
WAN	Wide Area Network
IETF	Internet Engineering Task Force
WWW	World Wide Web
DC	Data Compression

UA User Agents

UAC User Agent Client

UAS User Agent Server

RURI Uniform Resource Indicator

SDP Session Description Protocol

ITU International Telecom Union

IANA Internet Assigned Numbers Authority

DTLS Datagram Transport Layer Security

IWF Interworking function

TOPS Telephony System Packet

DOSA Distributed Open Signaling Architecture

QoS Quality of Services

SS7 Signaling System 7

MMT Multimedia Multiparty Teleconferencing

GCCG Generic Conference Control Gateway

CG Conference Gateway

URI Uniform Resource Locator

MOS Mean Opinion Score

GET LALUAN PERSIDANGAN MASA NYATA BAGI KLIEN HETEROGEN: KLIEN RSW (REAL TIME SWITCHING) DAN KLIEN IAX (INTER-ASTERISK EXCHANGE)

ABSTRAK

Pelbagai organisasi mengambil kira pengisyaratan bunyi dan video melalui Protokol Internet (Internet Protocol, IP) dengan pendekatan yang berbeza. Protokol IAX (InterAsterisk eXchange) digunakan sebagai protokol VoIP oleh penyedia perkhidmatan (service provider) kerana sifatnya yang ringkas dan mesra NAT. Sementara itu, pensuisan masa nyata (Real time SWitching, RSW) mampu menggabungkan perkhidmatan suara dan video. Namun demikian, disebabkan kedua-duanya mempunyai kebaikan dan keburukan yang tersendiri, ia menyukarkan pengguna untuk memilih penyelesaian yang terbaik. Dalam kebanyakan bidang, RSW digunakan. Namun demikian, IAX pula digunakan dalam kebanyakan penyedia VoIP. Oleh itu, kebolehoperasian bersama RSW dengan IAX dianggap penting dalam memaksimumkan pulangan pelaburan semasa, yang membolehkan RSW digunakan sebagai paket alternatif protokol isyarat telefoni.

Kami mencadangkan serta melaksanakan laluan get persidangan (conference gateway, CG) yang memudahkan klien heterogen berkomunikasi. Ia terdiri daripada dua modul, iaitu: isyarat dan media. Dalam modul isyarat, kami menukar isyarat dan mengawal mesej RSW dan IAX untuk mencapai komunikasi isyarat di antara klien heterogen melalui CG. Modul isyarat memasukkan keautentikan klien heterogen (RSW dan IAX) sebelum mereka mula berkomunikasi, menukar petunjuk sumber seragam (uniform resource indicator) untuk mencapai klien yang

jauh, menetapkan panggilan media dan menukar kebolehan media. Media masa nyata dapat diterjemahkan sebagai modul bagi penukaran data media klien (RSW dan IAX) melalui CG. Kami menguji komponen CG seperti isyarat dan media dalam senario yang berbeza. Bagi komponen isyrat, kami mengukur kelengahan atau kelambatan sesi yang ditetapkan di antara klien heterogen melalui CG. Bagi modul media, kami mengaruh (induced) pelbagai variasi rangkaian melalui CG. CG yang dicadang dan dilaksanakan memberikan hasil yang menyakinkan serta berkesan.

REAL TIME CONFERENCE GATEWAY FOR HETEROGENEOUS CLIENTS: REAL TIME SWITCHING CLIENTS AND INTER-ASTERISK EXCHANGE CLIENTS

ABSTRACT

Various standards organizations have considered signaling for voice and video over Internet Protocol(IP) from different approaches. The Inter-Asterisk eXchange (IAX) protocol is used as the promising VoIP protocol by the service provider because of its simplicity and NAT friendliness. Meanwhile, the Real time SWitching (RSW) has the ability to combine voice and video services. Incidentally, these two heterogeneous clients pose considerable problems for users who have to choose between two solutions offering different advantages and disadvantages. While RSW is being used in many areas, IAX is being deployed in many VoIP services. Hence, RSW interoperability and coexistence with IAX is considered very important to support new deployments that could use RSW as an alternative packet telephony signaling protocol.

We proposed and implemented the conference gateway (CG), which enables the heterogeneous clients to communicate seamlessly. It has two modules, one is signaling, the other is media module. In signaling module, we have converted signaling and control messages of RSW and IAX respectively to achieve seamless signaling communication between heterogeneous clients via CG. The signaling module includes authentication of heterogeneous clients (RSW and IAX) before they start communication, conversion of Uniform Resource Indicator to reach remote clients, setup media calls and exchange media capability. The real time media translation is the module which takes care of conversion of media data between the clients

(RSW and IAX) VIA CG. We have tested CG's components such as signaling and media in different network scenarios. For signaling components we have measured the session setup delays between the heterogeneous clients (RSW and IAX) VIA CG. For the media module we have induced the various network impairments to the CG's network. The proposed and implemented CG has shown promising results and encouraged to deploy it in wider scale.

CHAPTER 1

INTRODUCTION

1.1 Introduction

Clearly, the mission of media conferencing and collaborative computing is not only to bring individuals together, but also to make groups more effective at their work (Schooler, 1996). Designs for real time media communication control protocols were proposed in (Anerousis et al., 1999; Goyal et al., 1999; Huitema et al., 1999; ITU-T, 1996; Kalmanek et al., 2000; Rizzetto and Catania, 1999; Srinivasan et al., 2000; Willebeek-LeMair and Zon-Yin, 1997) to provide multimedia communication services over Internet Protocol (IP) based networks. IP based communication technology enables the deployment of multimedia applications combining a variety of media data, such as text, audio, graphics, images, and full-motion video.

Voice over IP (VoIP) is a technology to transport voice over IP networks. This technology connects people via packet switched networks instead of traditional circuit switched networks. Because of the low cost of Internet usage, VoIP can make telephone calls much cheaper than traditional public switched telephone networks (PSTN). Universities, enterprises, businesses, and corporate entities have also invested in the development and utilization of VoIP systems. Therefore, VoIP has attracted an increasing number of customers who previously opt for conventional communication solutions; as such, it is expected that VoIP will ultimately replace the PSTN (Conti, 2004).

The Real time Switching (RSW) control protocol was designed and developed in 1993 as a control mechanism for multimedia conferencing by the Network Research Group in School of

Computer Sciences, University Sciences Malaysia (USM). A multimedia conferencing system (MCS) uses the RSW control protocol to conduct audio and video conferencing. The RSW control criterion is used for establishing multimedia conferencing sessions in an IP network. A session could be a simple one-way media conference, or a collaborative multimedia-based and multipoint-to-multipoint conference session. The RSW Control Criteria was designed to reduce bandwidth requirements (Sureswaran and Abouabadalla, 2002) when many people use the RSW system; it also prioritizes participants to avoid confusion (i.e., when everybody speaks during a conference).

InterAsterisk eXchange (IAX) is a simple and complete VoIP protocol that can handle most common types of media streams. Lightweight and VoIP-friendly, it consumes less bandwidth than any other VoIP protocols (IETF, 2010). IAX is a binary protocol designed to reduce overhead, especially with regards to voice streams (IETF, 2010). It also multiplexes signaling and multiple media streams over a single User Datagram Protocol (UDP) stream. By using UDP over a single Internet port (Port 4569), both signaling and media are transmitted and received. IAX easily traverses firewalls and uses much less overhead than real-time protocol (RTP) (IETF, 2010). Considered as a simple protocol, it has the ability to avoid network address translation (NAT) traversal complications. There are many other features that IAX could offers that are unavailable in existing VoIP signalling protocols (Boucadair, 2009). For multiple calls, IAX reduces the overhead of each channel by combining data from several channels into one packet, thus reducing not only the number of headers, but also the number of packets.

The Research and Development activities on the VoIP and multimedia control protocols are on the rise. There are many services and applications offered at the endpoints. This kind of development and services on the endpoints, the traditional carriers should embrace the paradigm shift and use it to their benefit. The traditional carriers can offer scalable and very reliable services by leveraging the capabilities of the multimedia endpoints.

1.2 Research Problem and Direction

VoIP and multimedia control protocols networks have come of age, with significant investment and research and development from key industry players. New multimedia control protocols and network and communications technologies are dramatically changing the way services are deployed. Now a critical need for services that span heterogeneous networks, and multimedia control protocol (Rizzetto and Catania, 1999) (RSW, SIP, and H.323), with its extreme flexibility and widely deployed protocols, will be a key feature in enabling the communication for heterogeneous communication. With VoIP and multimedia control protocols, communications services are more logically implemented in centralized than in endpoints service components. This is because centralized media servers are far more matured, powerful and intelligent than endpoints and provide functionality (Isenberg, 1998).

VoIP, as IP telephony, has technologically matured and currently gives service providers opportunity to offer both traditional services, apart from a wide range of new multimedia conferencing applications, and exploit the power of sophisticated user terminals (Corrocher, 2003) . While services continuously provided for conventional VoIP, potential interaction with networks could further increase and enhance capabilities of multimedia conferencing systems. Yet, the VoIP system poses substantial challenges. The VoIP architecture must be able to support both traditional services and new services enabled by real-time multimedia conferencing endpoints. Real-time multimedia communications imply interactive communications can use various media (i.e., audio-conferencing and video-conferencing using IP networks). By leveraging the capabilities of multimedia conferencing system, VoIP service providers can offer multiple levels of services based on different levels of communication capabilities.

H.323 document specifies the requirement for multimedia communication over IP networks, which includes audio, video and data conferencing. It is unified system for performing these functions, leveraging the strengths of the ITU-T protocol. Many researchers have analyzed H.323 protocol and concluded that H.323 is more complex in several aspects (Basicevic et al., 2008). It also uses many protocols and so as many ports to be opened in the firewall to enable smooth trafficking of media. International Engineering Task Force (IETF) saw these problems in H.323 and they realized that the multiprotocol concept of multimedia session is not a suitable idea to have remote multimedia sessions. IETF then they released their protocol called Session Initiation Protocol (SIP). Unlike H.323, SIP takes the help of other protocols to create remote media sessions. SIP facilitates the formation, modification and execution of communication session between two or more participants. The participants can either be video-conferencing client or voice mail server. The protocol's roots extend back to 1996 and predate the widespread emergence of IP communication (Malim, 2002). SIP traces its roots to several IETF initiatives, and is tightly linked to World Wide Web (WWW), e-mail technologies and standards (Grobel, 2002). SIP's main advantage is that it is a short, simple and flexible protocol, which can run over most wired and wireless networks (Tanner, 2002).

To provide seamless connectivity between heterogeneous (H.323 and SIP) protocols the interworking or centralized conference gateway is needed Ho et al. (2001). But this concept of interworking also posed several problems such as duplication of signaling messages between H.323 and SIP VIA centralized conference gateway (De Marco et al., 2006).

RSW provides the solution of multimedia communication with control criteria. At the same time IAX protocol was released to cater the need of single protocol to perform VoIP sessions. Unlike SIP and H.323, the IAX protocol uses single port and protocol to perform remote media sessions (Boucadair, 2009).

The presence of VoIP and control criterion protocols has preordained that no single protocol will dominate the multimedia communication systems for a long time to come (Stephens and Cordell, 2001). Furthermore, service providers of multimedia communication network realize that people wish to communicate with each other irrespective of the multimedia communication service provider and protocol practiced on their IP network! A key to the future growth of IP based multimedia conferencing is the interworking of multimedia protocols which allows a seamless, end-to-end connectivity between all types of VoIP protocols and multimedia conferencing protocols (Radvision, 2001). The main goal of this thesis is to provide a bridge between RSW and IAX.

1.3 Research Scope

IAX is currently the industry's predominant standard for VoIP service over internet (Boucadair, 2009); but the RSW, in conjunction with the audio and video services, continues to generate significant interest in the market-place. The two control criterion address similar requirements but use different methods of addressing, locating end-points, call establishment and media negotiation. With its greater maturity, IAX based VoIP services continues to be deployed in operational networks, but a number of significant RSW services are likely to be deployed soon. However, even if RSW does become widely deployed, service providers may choose to maintain existing services (RSW) deployments in order to serve the needs of existing customers and applications. Furthermore, it may be found that some applications are better suited to IAX while others are better implemented with RSW. Also, some service providers may not want to change over to a new control criterion because it does not offer the features that their existing protocol does. For these reasons it is unlikely that one protocol will be completely dominant at a global level, and thus the mix of protocols used across administrative domains is likely to require interworking solutions well into the future.

The IAX protocol has been available for a few years. Meanwhile, the popularity of RSW has since increased due to its ability to easily combine voice and video services. Incidentally, these two heterogeneous clients pose considerable problems for users who have to choose between two solutions offering different advantages and disadvantages. While RSW is being used in many areas, IAX is being deployed in many VoIP products. Hence, RSW interoperability and coexistence with IAX is considered very important to support new deployments that could use RSW as an alternative packet telephony signaling protocol. Thesis uses IAX as opposed to Session Initiation Protocol (SIP), which although started as a simple and attractive for VoIP, has become a complex and heavy protocol to implement. Similarly, H.323 is a very complex protocol suite that can result in the transmission of many unnecessary messages across the network (Goode, 2002).

1.3.1 Why RSW to IAX translation protocol

Convensional (VoIP protocols such as SIP and H.323) are used as the main VoIP signaling session protocol in various organizations, service provider and VoIP equipment manufacturer, such as IP Multimedia Subsystems (IMS) and Telecoms and Internet converged Services and Protocols for Advanced Networks (TISPAN) architectures. This move is motivated by the simplicity of the protocols and its popularity within the IETF community. SIP and H.323 were the answer from the IETF community to the problem of specifying a protocol suitable for controlling multimedia sessions over packet based network. SIP has been used by service providers, becuase of its simplicity and its flexibility. Moreever, it suffers from several drawbacks, such as (Boucadair, 2009):

- Complications when crossing Firewalls.
- SIP uses RTP for media sessions, due to the dynamic RTP port-assignment policy.
- Complications are arised becuase of the path decoupled behaviour of SIP. Moreever, SIP service providers need to put an intermediate node in between the signaling and the

media path for access control.

- Complication when other protocols are deployed along with the SIP.

IAX is an interesting alternative besides conventional VoIP protocols, deployed nowadays by service providers for their VoIP service offerings (e.g. H.323 and SIP). The IAX protocol offers significant features unavailable in other existent VoIP signaling protocols. Apart from its simplicity, the benefits of the IAX protocol over other VoIP protocols are listed below (Boucadair, 2009).

1. IAX uses UDP as its transporter for media and signaling sessions and uses a single port, default IAX port is 4569.
2. The IAX registration is very much similar to the SIP one. An IAX client should contact a IAX server for registration by sending specific messages. Registration information is then retrieved by the IAX server and stored in its system within a time period.
3. IAX mixes the signaling and media paths. The seperation of media and signaling is possible once the connection has been successfully established.
4. IAX is not depending on other protocols for media transmission. IAX handles media streams itself. IAX supports any media types such as voice, video, image text, and HTML.
5. IAX has defined reliable and unreliable messages. Unreliable messages are media data. Which are not acknowledged or retransmitted if they are lost in the network. Reliable messages are signaling sessions, becuase IAX identifiers maintained by the IAX clients during signaling session. These messages should be acknowledged by the peers; if not, these messages are retransmitted to peers.

6. NAT traversal is not a big issue with IAX. IAX's siganling message never carry IP addresses.

7. IAX measures the network performance and these measured information can be shared with other IAX clients during an active session.

8. IAX client will be notified about its peers.

9. IP security modules can be deployed together with IAX. IAX entertains exchange of shared keys. It may be used either with plain text or in conjunction with encryption mechanisms like Advanced Encryption Standard (AES).

10. IAX authentication is implemented becuase 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.

The idea of physical round table meeting is implemented in the RSW Control Criteria. The RSW Control Criteria is focused more on badwidth reduction when a lot of people using the RSW system and priortizing the participants to avoid confusion when every body speaks up during confrence. In any round table meeting or multimedia conference is made up of a conference chairman, participants and passive observers. The chairman person of the conference is the organizer of the conference, while other conference members can be participants or observers (Sureswaran and Abouabadalla, 2002). Following items make RSW as one of the best control criteria for media conferencing on IP.

- Chairman only controls the conference.
- Has the ability to change CODEC of audio and video during the conference.
- Seperate streams for Audio and Video real time data.

- Has the ability to cross any NAT device.
- Has the ability to perform multiserver conferences.

1.4 Objectives

The objective of this thesis is to bridge heterogeneous clients (RSW and IAX) through a common Conferencing gateway (CG). Therefore, following are the objectives.

1. Comparative analysis of media setup protocols (IAX and RSW), to determine common and unique features.
2. To develop a messaging mechanism for interoperability between IAX and RSW in terms of: signaling, registration, call setup and capability negotiation.
3. To develop procedures for interoperability between IAX and RSW in terms of real time data transmission.
4. To evaluate performance of the interoperability schemes in terms of signaling session setup and real time media translation.

1.5 Research Methodology

In order to achieve the objectives mentioned in the above section, the research process is organized using the following steps:

- Review and evaluation of the existing methods.

  In this phase of research, we do literature survey on Media Communication Systems, VoIP systems and Inter-Working frameworks. Most recent literature on interworking system have been studied.

- Conceptual design.

  In this phase of research, we focus on designing the conceptual architecture of the proposed solution.

- Implementation, experimentation and quantitative evaluation.

  Lastly, we implement a prototype of the proposed Conference Gateway and setup two experiments with different purposes. The first one is to evaluate and validate signaling sessions of IAX and RSW VIA CG. The second experiment deals with real time media transmission between IAX and RSW VIA CG and will be measured in the presence of network packet dealy, network packet loss and packet reodering.

1.6 Thesis Contribution

In what follows, summaries of thesis contributions to achieve seamless connectivity of signaling and media between heterogeneous clients:

1. A comparative analysis of the media setup protocols (IAX and RSW) in terms of signaling sessions, media capability negotiation and real time data transmission is performed.

2. A mechanism to translate each and every signaling messages of the heterogeneous clients (IAX and RSW) is developed to achieve registraion, call setup and media negotiation between IAX and RSW.

3. Procedures are developed to achieve seamless real time media transmission between hetergeneous clients (RSW and IAX).

4. Evaluation is perfomed to measure the signaling mechanism in terms of session set up. Precedures are also evaluated to measure the media translation during network impairments such packet delay, packet loss and packet reodering.

Currently RSW is the most widely used protocol for PC-based conferences (Abouabdalla and Sureswaran, 2006), while carrier networks using IP telephones seem to be built based on IAX. IAX and RSW protocols both provide mechanisms for call control. Interworking between the two protocols is desirable in order to achieve universal connectivity. Interworking will include two types of endpoints: IAX terminals and RSW endpoints (end user RSW clients). Other major entities include RSW and IAX Interworking Function (RI-IWF). RSW is not as strictly defined as a complete system as IAX. Many aspects of the IAX architecture are left open to interpretation. IAX can integrate with other Internet protocols, such as the Media Gateway Control Protocol (MGCP), SIP and H.323 to constitute a complete system.

1.7 The Organization of the thesis

Chapter 2 gives an overview of protocols used to control multimedia conferencing systems. This leads to a study on RSW control criteria architecture and IAX architecture. It also gives a brief of interconnection between various other protocols. A summary of the major problems in such protocols is given, followed by an indication of the proposed solutions. As a means to put the main proposal into proper perspective (which is the communication translation protocol to be given in chapter 4) the two protocols (RSW and IAX) components are given, followed by a presentation of why the need of connecting the two different clients or protocols.

Chapter 2 also gives a detailed description of IAX protocol standard. This description includes the structure of IAX protocol, IAX requests and responses and IAX message format. This will lead to a study on all IAX message processes, transactions and error handling, which would provide the second part of the benchmarks for the proposed communication translation protocol. A summary of RSW and IAX is also given in this chapter. **Chapter 2** gives a detailed description of RSW control criteria, its options and the options adopted to develop a complete multimedia conferencing system. This naturally leads to a study on distributed network entities architecture and entities required by RSW control criteria. It also leads to a study on message formats involved in RSW based conference, which would provide the first part of the benchmarks for the proposed communication translation protocol. **Chapter 3**, presents the comparative analysis of IAX and RSW under different network conditions. It also presents the complete architecture of the CG and its components. **Chapter 4**, provides the validation of the signaling modules of the CG server.

Chapter 5, in this chapter we have presented media module of the CG server and testing of media modules of the CG. We have induced network impairments to the network of the CG and measured the performance of the CG under difference condintions (impairments). **Chapter 6** presents a summary of the discussions. An outline of possible future work that can be done to enhance and continue the research is also discussed here.

CHAPTER 2

LITRATURE REVIEW

2.1 Introduction

The IP based audio and video communication domain has matured beyond providing VoIP only services. Universities, enterprises, businesses, corporate and individual consumers often use VoIP. The focus of IP based communications is now on the operations of the media system (Smith et al., 2003), management of the media system, administration and provisioning aspects of large-scale media servers (Ventakesha Prasad et al., 2005; Prasad et al., 2005) and reliable and secure communication systems (Chou, 2007). As IP based communications system grapples with these issues, newer technologies in the form of cloud-based IP communication systems; peer-to-peer VoIP networks (Fiedler et al., 2006); interworking of VoIP and multimedia control criteria; use of IP communications in virtual worlds; social networks and IP communications are starting to assert a strong presence in the IP communications domain. In such a demanding scenario, IP based multimedia communication should be, error free system, and need to remain flexible and scalable.

This chapter provides an overview of existing multimedia conferencing control protocols or standards and related work on interworking of protocols. It also shows the problems faced by such protocol in general and an indication of the solutions required. This chapter also presents IAX standards and the RSW control criteria which we shall adopt in the design of the communication translation protocol. This will be the base for our proposed study.

2.2 Protocols to control multimedia conferencing sessions

A protocol is defined as a specific set of rules, procedures, or conventions relating to the format and timing of data transmission between two or more devices (Newton, 2003). For a multimedia conferencing or IP telephony to be able to work in today's network, a form of control mechanism (protocol) should be used. This mechanism should provide lightweight means for creating and ending sessions for real-time multimedia communications over IP based networks mainly for voice and video, but also for videoconferencing, chat, and application sharing. Following are the mostly used multimedia session control protocols.

- Real Time Switching (RSW) control Criterion
- Session Initiation Protocol (SIP)
- H.323
- InterAsterisk eXchange protocol (IAX)

2.3 IP communication and Distributed network entities

An IP-based communication system is divided into smaller programming modules, with every module interacting with each other using a certain set of rules. All these modules are placed in a distributed architecture that involves real-time constraints and requirements. Moreover, non-functional requirements, particularly time-dependent properties, also play an important role. IP-based communication requires grouping a large number of distributed network entities in a resonant fashion in order to provide the required set of services. Integrating a large number of distributed network entities leads to the possibility of failure at various legs, which could affect the availability and performance of the system. So these entities are created individually with specific functions and are implemented based on the following design principles:

- Each network entity is independent of each other.
- Each network entity communicates with each other's using the Internet Protocol (IP) or other means of communication protocols within the system.
- Each network entity should be presented in a black box style format, where its input parameters, its functions using input parameters and output parameters of the black box must be defined.
- Each network entity can independently be plugged into a network and perform its functionalities or can be unplugged without affecting the system.

Implementation norms state that distributed network entities have to follow the Open Systems Architecture (a requirement for using non-proprietary hardware). Furthermore, these entities should also be implementable using any operating system that can support the given specifications. With such norms, network entities should be able to function anywhere on the given network (Sureswaran, 1998; Tharmaraj, 1999).

A distributed multimedia conferencing environment is also based on IP networks; each of its modules has a specific form of functionality. The modules can work independently and also they can perform their functions in a group. Some of these modules in conferencing system are based on bandwidth reduction. And these modules are plugged into the network to provide more efficiency in terms of bandwidth reduction or any specific function related to the IP-based communication. "Unplugging" them from the network does not cause the network or the system to crash or to exhibit any unwanted behavior. As good example of this module, if you plug in a Data Compression (DC) module to the network, the conferencing system's bandwidth usage will go down; when you unplug this entity, the system will not crash even though the system will be using more bandwidth.

The distributed communication system are designed and deployed according to the requirement and specification. In RSW, four distributed network components are in use to support the Multimedia Conferencing Application. They are categorized as server entity, client entity, multi LAN IP converter (MLIC) entity, and data compression (DC) entity. These entities will be described in more detail later in this chapter.

For the RSW distributed entities to be well accepted and supported in today's network environment of multiple protocols, various operating systems, and different network architectures, a management entity is required. This management entity can be carried out by the control criteria, enabling the entire communication system to work as a cluster.

2.4 RSW Control Criteria

Most of the Multimedia conferencing systems try to provide real-time connections as well as receive and transmit capabilities (Anupam and Bajaj, 1993, 1994). This means all or most of the participants are decoding the received real time data and broadcasting the real time data at the same time (Altenhofen et al., 1993; Buford, 1994; Ishii and Miyake, 1991). This would be considered an ideal situation if not for these two factors:

- The confusion generated when everyone tries to speak at the same time.
- The tremendous amount of network traffic generated by all these participating sites.

The RSW conducts communication based on the round table meeting. Thus, to resolve the first issue, one simply has to observe how an actual conference is conducted around a table.

The second issue could be addressed based on the control method used to resolve the first issue. The bandwidth utilization cannot be removed completely. In any round table meeting, only one person is allowed to talk at any given time. Thus, making use of the situation into a multimedia conferencing system, the huge amount of network traffic can be reduced by allowing only one station to transmit at any given time.

The RSW Control Criteria is a set of control options used to control multimedia conferences (Sureswaran and Subramanian, 1995; Sureswaran, 1994). The control criteria provides the basis for optimizing multimedia (video, audio, and document) conferencing bandwidth requirements, as well for maintaining a form of control and order over the conference (Sureswaran, 1996). The control criteria is based on the following two fundamental principles:

- The need for a control crietria to coordinate with active conference.
- Only one station may broadcast or make changes (collaborative document conferencing) at any given time.

Both fundamental principle are focused on reducing the unnecessary network bandwidth by inducing a control mechanism to maintain order among involved conferencing participants, and only one active site can be allowed to broadcast at any given time and while all other passive sites are allowed passive participation only. Thus, bandwidth requirements remain the same even though the number of participant stations increases.

Based on the above issues, the RSW Control Criteria was created. Thus, RSW provides more benefits, including the following:

- The ability to hold real-time multimedia conferences like Desktop Video Conferencing over long delay networks (satellite links).
- The ability to coordinate changes being done to a shared document because these changes must be done sequentially in order to avoid conflicts or errors (data integrity problem).

- The ability to support a distributed processing environment for multimedia conferencing.

The RSW Control Criteria, first of all, defines the status of sites involved in a conference. The RSW gives sites different access and control privileges over the conference. It assigns site status during the initiation of the conference. On the fly a site's status can be changed dynamically during the conference.

2.4.1 Site status within RSW control protocol

The RSW Control Criteria provides six different types of client sites status and one server site is involved in a multimedia conference (Sureswaran, 1998). Figure 3.1 shows the relationship between the different sites.

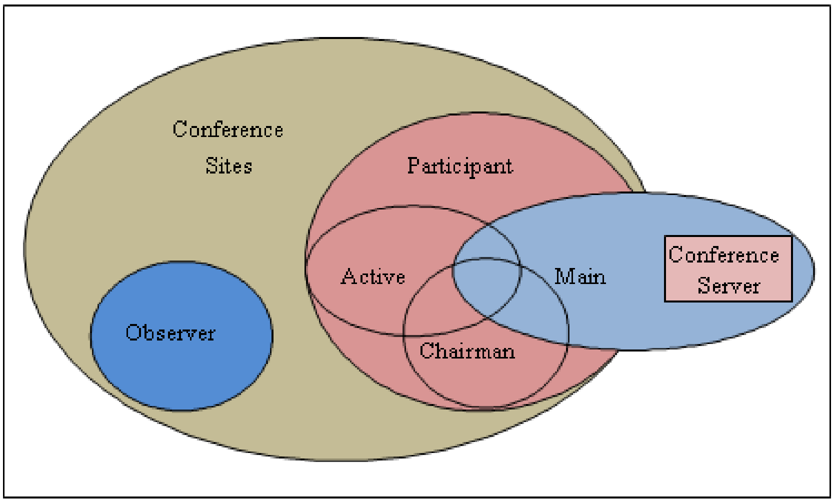

Figure 2.1: Site Status Relations

The Conference sites are all the sites involved in the conference, which are; the main site, active site, participant site, chairman site and observer site. The Observer site is a site that is allowed to be passively view the conference and is not allowed to take active part in the conference.

The Participant site, who can actively participate and contribute towards the conference. In other words, a participant can request and become an active site. A participant site can also be given main site status, depending on the RSW Control Criteria that is being used within the conference. However, the main site status must always be a participant site.

The Chairman site is the site that initiates the conference. The chairman site will have main site status if the RSW Control Criteria used is the option in which the chairman gets to decide who the next active site will be (Chairman Main Site Option). The chairman site, however, is the only site that will always have the following features:

- The ability to cut short a lengthy active site, that is, to "kill" a currently active site.
- To become the default site if there is no other active site requests.
- To have the ability to change the status of a participant to observer and vice-versa.
- The ability to end or terminate the conference.

There can only be one chairman site per conference. The chairman site is considered a static site and cannot be reassigned to another participant site during that conference.

The main site is the main controlling site for the conference. The main site is automatically assigned to the conference chairman site during the start of most conferences.

An active site is the only site in the conference that is allowed to transmit its multimedia stream onto the network. The active site is dynamic and when a site is designated to become the active site, it is the only site that is allowed to make any changes to the conference's audio, video and document.

A participant becomes an active site by making an active site request, then awaiting his or her turn to be given the active site status. The active site has the following unique features:

- There can only be one active site at any one time in any one conference.
- The active site is the only site that can transmit the multimedia stream onto the network.
- The chairman becomes the default active site if there are no requests for active site status.

The Conference server is not really a site, but a server that manages the conference. A client will have to first connect (log into) a conference server before it can begin a audio and video conference. The server will establish a virtual point to point TCP link to all clients to handle control communications and another link to handle document communications.

2.5 RSW Control Criteria Control Options

RSW control criteria are equipped with six different options for controlling a multimedia conferencing system. Any combination of these options can be used to control a conference as long as no contradictions arise. Furthermore, a conference is made up of a conference chairman, participants and observers. The conference chairman is the organizer of the conference, while other conference members can be participants or observers (Sureswaran and Abouabadalla, 2002).

2.5.1 Option 1: Equal Privileges

With this option, all conference sites have an equal opportunity of becoming active sites. The user that gets active site status is also given main site status and the privilege of choosing the next active site. The user that gets active site status is also given main site status and the privilege of choosing the next active site.

2.5.2 Option 2: First come first serve

This option can be applied to make decision for choosing the active sites resides within the server. The RSW will assign active site status to the sites in the order the request comes in.

2.5.3 Option 3: First come first serve, with time-out

This option is similar to option 2, except that there is a time-out feature incorporated within the application itself. Each site is only allowed a certain maximum time limit for broadcasting, reaching this limit; the site is warned that they will automatically be cut off within the next x seconds.

2.5.4 Option 4: Organizer Main site

With this option, the RSW gives the privileges of choosing the active site to the site that organizes (Chairman) the conference, which will then hold the status of main site. Thus now, the conference organizer will decide which site is to be the next active site.

2.5.5 Option 5: Restricted Active sites

In this option, the organizing (chairman) site will act as an access pacifier for the sites allowed to participate in the conference. The participating sites may be that of a mutual prearrangement made earlier by the organizers of the conference. Other sites may be allowed to participate only as observers and never given active site privileges. This option is not used alone, but is used to enhance one of the existing options.

2.5.6 Option 6: Restricted active sites, upgradeable observer sites

This option is very similar to that of option 5, except that we will try to resolve the main limitations of option 5, which is the disability to upgrade observer sites to active sites in real-time. If the main site control, which is the only site that can permit the upgrade, resides within the application, then the application will have to decide if the observer site requesting active site should be upgraded. If the application was to decide, then a decision taking algorithm will have to be used to help process the decision.

2.6 Distributed network entities to support RSW control criteria

RSW control criterions are applied to control multimedia conferences in a distributed environment. The basic building blocks of RSW control criteria are the client workstations and the conference server. The client workstation is completely independent of the server. These two workstation are further extended by adding the multilan IP address conversion unit and the WAN data compression unit. Each of these units are separate independent units. They function like black boxes. Figure 3.2 shows the general network architecture of the RSW entities and how 2 different IP networks can be interconnected over WAN.

The input and out parameters of each of these black boxes are defined. The input information style, language and operating systems are transparent to the user and even to the entire system. To support RSW control criteria, four units were defined as distributed network entities. They are:

- The server entity
- The client entity
- The multilan IP converter (MLIC) entity
- The data compression (DC) entity

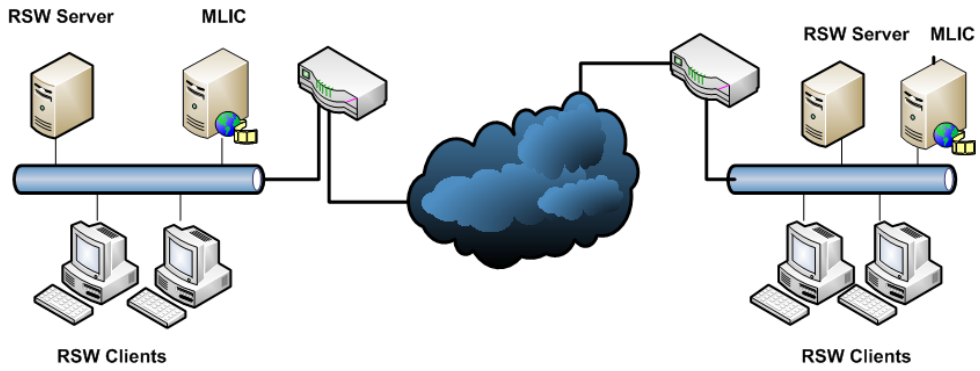

Figure 2.2: General RSW network architecture

2.6.1 The server entity

The server entity looks after the conferencing endpoints. Server entity keeps track of the all active users and passive users and it also handles the authentication and authorization of the participants. The functions of the server are as follows:

- Coordinates and manages all network entities involved in a multimedia conference.
- Uses RSW control criteria to control the conferencing.
- Providing users a platform to register/login to participate in conferences.
- Coordinates multicast address assignments for single and multiple conferences.
- Establishes inter-server links (only during multi-server conferences).
- Provides damage control when links break or when entities 'crash' (become 'unplugged').

2.6.2 The client entity

Figure 3.3, is the client entity and it is user based GUI application that works on the end user's site. The modules in the client entity are video module, audio module, signaling module, authentication module and conference monitoring module (Tharmaraj, 1999). The functions of the client are as follows:

- Multimedia captures (AUDIO and VIDEO units).
- Signaling Modules (To setup and tear down conference calls)
- Data packetization (for transmission) and data reconstruction (for receiving and playback).
- Authentication Module (RAS).
- Conference Monitoring unit to Communicate with the server to maintain the RSW Control Criteria.

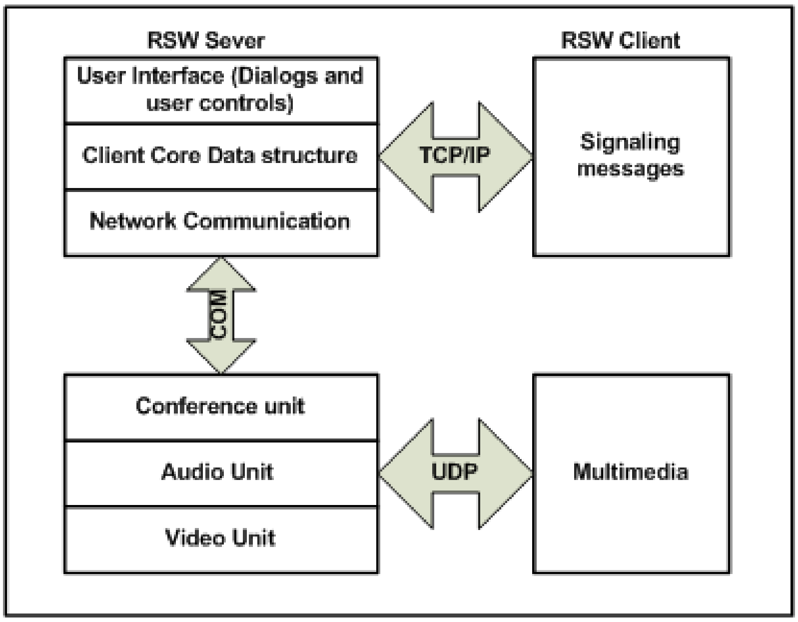

Figure 2.3: Client entity

2.6.3 The Multiple LAN IP Converter (MLIC) entity

Since the real time media packets (audio and video) transmitted as multicast packets, the WAN/LAN routers generally drop these packets. This means conference cannot be held between users from different LANs. Figure 3.4 shows the architecture with MLIC (Sureswaran, 1998).

MLIC entity is needed for the users of two different LAN to conduct audio and video conferencing. MLIC manages to convert multicast packets of audio and video data to the unicast to pass through the WAN routers. On the receiving end, another MLIC converts these unicast packets and retransmits them as multicast packets onto the other LAN (K. and Sureswaran, 2000). The functions of the MLIC can be defined as follows:

1. Audio/Video packets are transmitted by the client (active site) in LAN 1. MLIC in LAN 1 will:
    - Listen to specified port for Audio/Video UDP multicast packets.
    - Change to Audio/Video UDP unicast packets and transmit out.
2. The converted packets then go through the WAN router to LAN 2. The MLIC in LAN 2 will then:
    - Receive Audio/Video UDP unicast packets from the MLIC in LAN 1.
    - Change Audio/Video UDP unicast to Audio/Video UDP multicast packets and retransmit within LAN 2.