Università degli Studi di Napoli Federico II

Facoltà di Ingegneria

Dottorato di Ricerca in Ingegneria Informatica ed Automatica
XXIII Ciclo
Dipartimento di Informatica e Sistemistica

A standard path towards scalable conferencing in the Internet

Alessandro Amirante

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Tutor
Prof. Simon Pietro Romano

Co-tutor
Prof. Henning Schulzrinne

Coordinator
Prof. Francesco Garofalo

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Abstract

The work described in this Ph.D. thesis has been carried out within the context of computer networks and real-time multimedia applications over the Internet. Specifically, we focus on conferencing, a challenging service which experienced a wide growth during the last years. We examine the standardization efforts conducted in this field by the Internet Engineering Task Force (IETF), which are mainly focused on *centralized* conferencing as defined by the XCON Working Group. We actively contributed to such efforts by designing and implementing the defined framework and protocols, while also writing *best current practice* documents which are on the path to become *Request For Comments* (RFCs). The main outcome of such activities has been Meetecho, a standards-compliant multimedia conferencing and collaboration platform we developed. Meetecho has been conceived at the outset to be extensible towards a distributed architecture, yet being fully compliant with the XCON specification, in order to better fulfill scalability requirements resulting from its large-scale deployment. The distributed architecture, which is the subject of the DCON (Distributed Conferencing) proposal we submitted to the IETF, is thoroughly described herein, where we also provide the reader with the results of a scalability analysis we conducted in order to assess the actual performance improvement attainable with distribution. The figures obtained have been encouraging and definitely motivated us in pushing our proposal into the Internet standardization community. Finally, a remarkable part of this dissertation is focused on diagnosing and addressing issues that might arise when deploying multimedia architectures in the actual Internet.

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Chapter 1

Multimedia Conferencing

1.1 Introduction

Conferencing can nowadays be considered by providers as an extremely challenging service, since it imposes a number of stringent requirements to the underlying network infrastructure. First, the intrinsic multimedia nature of a conference (which typically involves a combination of audio, video, instant messaging, desktop sharing, etc.) requires coping with complex issues like session management and floor control. Second, the real-time features of conference-based communication call for an appropriate level of Quality of Service (QoS). This chapter presents the reader with standardization process associated with multimedia conferencing over IP the main international standardization bodies are fostering.

1.2 Background

The most widespread signaling protocol for IP networks is the Session Initiation Protocol (SIP) [30]. It provides users with the capability to initiate, manage, and terminate communication sessions. SIP natively allows multiparty calls among multiple parties. However, conferencing does represent a more sophisticated service that can be seen as an extension of multi-party calls where audio is just one of the possible media involved. For example, the conferencing service may provide video functionality as well as instant messaging, files and presentations sharing or even gaming. Furthermore, the conferencing service provides the means for a user to create, manage, terminate, join and leave conferences. Finally, it provides the network with the ability to deliver information about these conferences to the involved parties. Over the last few years, standardization efforts have been devoted to conferencing related matters by international bodies like the IETF and the 3GPP. The Internet Engineering Task Force (IETF) is an open international community concerned with the evolution of the Internet architecture and protocols. Within the IETF, the Centralized Conferencing (XCON) working group is explicitly focusing on multimedia conferencing. Furthermore, there is another working group whose standardization activity also dealt with conferencing related issues: the Session Initiation Proposal Investigation (SIPPING) WG. It developed the very first framework for multi-party conferencing based on the SIP protocol [27]. This framework defines a general architectural model, presents terminology, and explains how SIP is involved in a tightly coupled conference. It is the subject of Section 1.3. Taking inspiration from the work carried out in SIPPING and willing to release any constraint about the signaling protocol, the XCON WG worked hard on the definition of both a reference framework [5] and a data model [21] for tightly coupled conference scenarios, which are described in Section 1.4.

The 3rd Generation Partnership Project (3GPP) actually represents a collaboration agreement among a number of regional standard bodies, born with the main objective of developing Technical Specifications for a third-generation mobile system based on GSM. Recently, the 3GPP has worked on the specification of a tightly-coupled conferencing service. Both the requirements and the architecture for such a service have been defined [1]. The cited document indeed represents a sort of integrated specification within the IMS, aimed at harmonizing the combined use of existing standard protocols, like the Session Initiation Protocol (SIP), SIP Events, the Session Description Protocol (SDP) and the Binary Floor Control Protocol (BFCP).

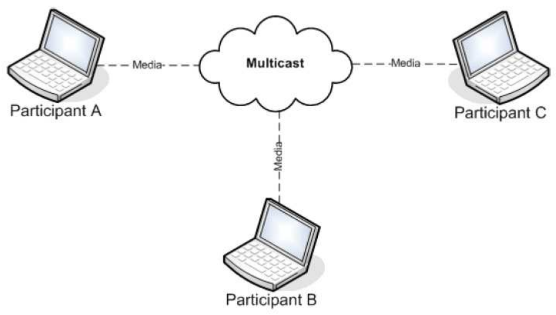


Figure 1.1: Loosely coupled conference

1.3 SIPPING Conferencing Framework

The SIPPING conferencing framework is the very first attempt the IETF made to standardize the multimedia conferencing service. The SIP protocol can support many models of multi-party communications. One, referred to as *loosely coupled conferencing*, makes use of multicast media groups (see Fig. 1.1). In the loosely coupled model, there is no signaling relationship between participants in the conference. There is no central point of control or conference server. Participation is gradually learned through control information that is passed as part of the conference (using the Real-Time Control Protocol (RTCP), for example). Loosely coupled conferences are easily supported in SIP by using multicast addresses within its session descriptions.

In another model, referred to as *fully distributed multiparty conferencing*, each participant maintains a signaling relationship with the other participants, using SIP. There is no central point of control, it is completely distributed among the participants (see Fig. 1.2).

Finally a third model, referred to as the *tightly coupled conferencing*, envisages the presence of a central point of control to which each participant connects to. It provides a variety of conference functions, and may possibly perform media mixing functions as well.

The SIPPING WG developed a framework for tightly coupled conference scenarios, presenting a general architectural model for these conferences and

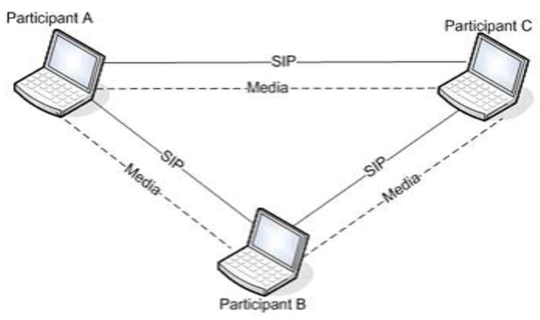


Figure 1.2: Fully distributed multiparty conference

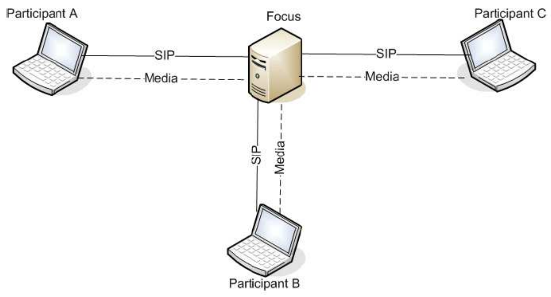


Figure 1.3: Tightly coupled conference

discussing the ways in which SIP itself is involved.

1.3.1 Overview of the architectural model

The central component introduced in the SIPPING architectural model is called *focus*, and maintains a SIP signaling relationship with each participant in the conference. The result is a star topology, as depicted in Fig. 1.3.

The focus is responsible for making sure that the media streams that constitute the conference are available to the participants in the conference. It does that through the use of one or more mixers, each of which combines a number of input media streams to produce one or more output media streams. The focus uses the media policy to determine the proper configuration of the mixers and has access to the conference policy, an instance of which exists for each conference. Effectively, the conference policy can be thought of as a database that describes the way the conference should operate. It is responsibility of the focus to enforce those policies. Not only does the focus need read access to the database, but it needs to know when it has changed. Such changes might result in SIP signaling (for example, the ejection of a user from the conference using `BYE`), and those changes that affect the conference state will require a notification to be sent to subscribers using the conference notification service. The conference is represented by a URI that identifies the focus. Each conference has a unique focus and a unique URI identifying that focus. Requests to the conference URI are routed to the focus responsible for that specific conference. Users usually join the conference by sending an `INVITE` to the conference URI. As long as the conference policy allows, the `INVITE` is accepted by the focus and the user is brought into the conference. Users can leave the conference by sending a `BYE`, as they would in a normal call. Similarly, the focus can terminate a dialog with a participant, should the conference policy change to indicate that the participant is no longer allowed in the conference. A focus can also initiate an `INVITE` to bring a participant into the conference. The notion of a conference-unaware participant is important in this framework. A conference-unaware participant does not even know that the User Agent (UA) it is communicating with happens to be a focus. As far as it is concerned, it appears like any other UA. The focus, of course, is aware of its duties, and performs the tasks needed for the conference to operate. Conference-unaware participants have access to a good deal of functionality. They can join and leave conferences using SIP, and obtain more advanced features through stimulus signaling. However, if the participant wishes to explicitly control aspects of the conference using functional signaling protocols, it must be conference-aware. A conference-aware participant is one that has access to advanced functionality through additional protocol interfaces, which may include access to the conference policy through non-SIP-specific mechanisms. The participant can interact with the focus using extensions, such as `REFER`, in order to access enhanced call control functions. The participant can `SUBSCRIBE` to the conference URI, and be connected to the conference notification service provided by the focus. Through this mechanism, it can learn about changes in participants, the state of the dialogs and the media. The participant can communicate with the conference policy server using some kind of non-SIP-specific mechanism by which it can affect the conference policy. The interfaces between the focus and the conference policy, and between the conference policy server and the conference policy are non-SIP-specific. For the purposes of SIP-based conferencing, they serve as logical roles involved in a conference, as opposed to representing a physical decomposition.

1.4 XCON: Centralized Conferencing

The purpose of the XCON Working Group and its framework is to achieve interoperability between the logical entities developed by different vendors for controlling different aspects of advanced conferencing applications. The SIPPING Conferencing Framework described in the previous section provides an overview of a wide range of centralized conferencing solutions known today in the industry. The logical entities and the listed scenarios are used to illustrate how SIP can be used as a signaling means in these conferencing systems. The SIPPING Conferencing Framework does not define new conference control protocols to be used by the general conferencing system and uses only basic SIP, the SIP conferencing call control features [16], and the SIP Conference Package [31] for simple SIP conferencing realization. On the other hand, the centralized conferencing framework specified by the XCON WG defines a particular centralized conferencing system and the logical entities implementing it. It also defines a particular data model and refers to the set of protocols (beyond call signaling means) to be used among the logical entities for implementing advanced conferencing features.

1.4.1 Framework

The XCON framework extends the SIPPING model by making it independent from the signaling protocol employed, while following the same principles and adopting the same terminology. Hence, the conference scenarios supported are *tightly coupled conferences*. In addition to the basic features, a conferencing system supporting the XCON model can offer richer functionality, by including dedicated conferencing applications with explicitly defined capabilities, along with providing the standard protocols for managing and controlling the different attributes of these conferences.

The centralized conferencing system proposed by the XCON framework is built around a fundamental concept of a *conference object*. A conference object provides the data representation of a conference during each of the various stages it goes through (e.g., creation, reservation, active, completed, etc.). It is accessed via the logical functional elements, with whom a conferencing client interfaces, using the various protocols identified in Fig. 1.4. Such functional elements are a *Conference Control Server*, *Floor Control Server*, any number of *Foci*, and a *Notification Service*. A Conference Control Protocol (CCP) provides the interface between a conference and media control client and the conference control server. For such purpose, the working group is specifying a dedicated protocol called Centralized Conferencing Manipulation Protocol (CCMP), which is briefly introduced in Section 1.4.2. A floor control protocol, instead, provides the interface between a floor control client and the floor control server. Section 1.4.2 touches on how the Binary Floor Control Protocol provides such feature. A call signaling protocol (e.g., SIP, H.323, Jabber, Q.931, ISUP, etc.) provides the interface between a call signaling client and a focus, while a notification protocol (e.g., SIP Notify [23]) provides the interface between the conferencing client and the notification service. A conferencing system can support a subset of the conferencing functions depicted in Fig. 1.4. However, there are some essential components that would typically be used by most other advanced functions, such as the notification service. For example, the notification service is used

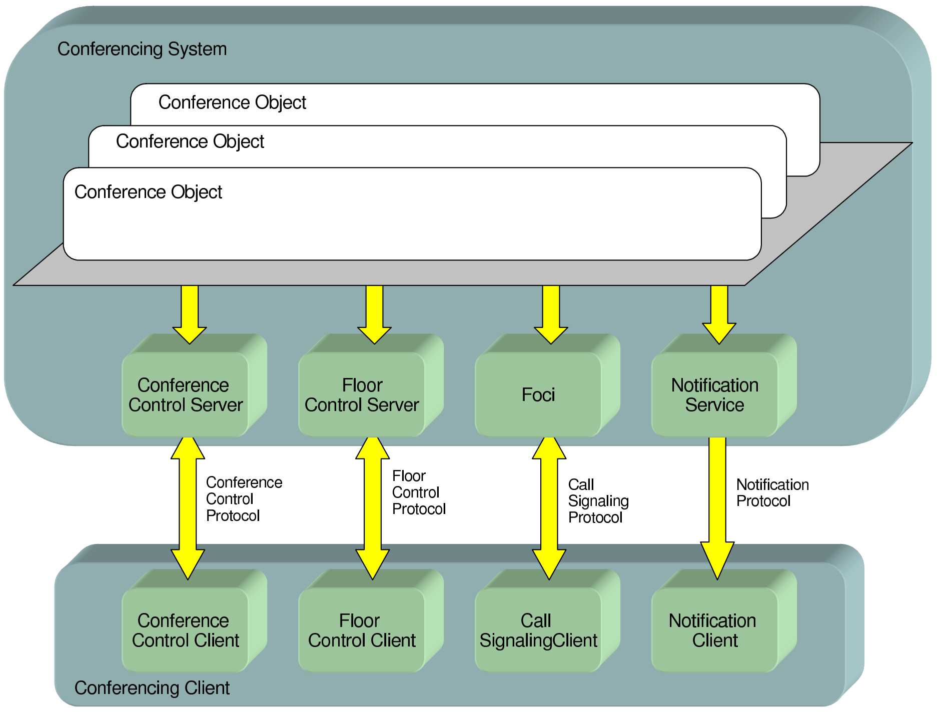


Figure 1.4: XCON framework logical decomposition

to correlate information, such as the list of participants with their media streams, between the various other components.

1.4.2 Dedicated protocols

Conference management

The latest output of the XCON WG is a protocol for the management and manipulation of the conference object: the Centralized Conferencing Manipulation Protocol (CCMP) [6]. CCMP is a stateless, XML-based, client-server protocol carrying in its request and response messages conference information. It represents a powerful means to control basic and advanced conference features such as conference state and capabilities, participants and relative roles and details. It allows authenticated and authorized users to create, manipulate and delete conference objects. Operations on conferences include adding and removing participants, changing their roles, as well as adding and removing media streams and associated end points. CCMP is based on a client-server paradigm and is specifically suited to serve as a conference manipulation protocol within the XCON framework, with the Conference Control Client and Conference Control Server acting as client and server, respectively. The CCMP uses HTTP as the protocol to transfer requests and responses, which contain the domain-specific XML-encoded data objects defined in the XCON data model [21].

Floor control

Floor control is a way to handle moderation of resources in a conference. In fact, a floor can be seen, from a logical point of view, as the right to access and/or manipulate a specific set of resources that might be available to end-users. Introducing means to have participants request such a right is what is called "floor control". A typical example is a lecture mode conference, in which interested participants might need to ask the lecturer for the right to talk in order to ask a question. The Binary Floor Control Protocol (BFCP) [10] has been standardized by the IETF for such purpose. This protocol envisages the above mentioned floor as a token that can be associated with one or more resources. Queues and policies associated with such floors are handled by a *Floor Control Server (FCS)*, which acts as a centralized node for all requests coming from *Floor Control Participants (FCP)*. Decisions upon incoming requests (e.g., accepting or denying requests for a floor) can be either taken on the basis of automated policies by the FCS itself, or relayed to a *Floor Control Chair (FCC)*, in case one has been assigned to the related floor. These decisions affect the state of the queues associated with the related floors, and consequently the state of the resources themselves. Considering again the lecture mode scenario example presented before, a participant who has been granted the floor (i.e., the right to ask a question to the lecturer) would be added to the conference mix, whereas participants without the floor (or with pending requests) would be excluded from the same mix, thus being muted in the conference.

Chapter 2

Meetecho: a standard multimedia conferencing architecture

2.1 Introduction

In this chapter we present a conferencing architecture called *Meetecho*. To the purpose, we embrace a practical approach, by describing an actual implementation of an open source centralized video-conferencing system capable to offer advanced communication experience to end-users through the effective exploitation of mechanisms like session management and floor control. Meetecho has been designed to be fully compliant with the latest standard proposals coming from both the IETF and the 3GPP and can be considered as an outstanding example of a real-time application built on top of the grounds paved by the SIP protocol. We will discuss both the design of the overall conferencing framework and the most important issues we had to face during the implementation phase.

2.2 Design

We started our design from an architectural perspective of the service we wanted to achieve, that is an advanced conferencing application. The first step was obviously identifying and locating all the logical elements which would be involved in the above mentioned scenario. We then investigated the possibility of replicating, or at least replacing, such elements with existing real-world components.

First of all, the scenario clearly addresses two distinct roles, a server side (the elements providing the service) and a client side (all users accessing the service). We referred to these roles in identifying the elements. At the client side, the very first mandatory element that comes into play is the *User Equipment (UE)*, which has to be both SIP-compliant and XCON-enabled in order to correctly support conferencing. On the server side, instead, we identified different cooperating components: the *Application Server (AS)*, the *Media Server (MS)* and one or more *Gateways*. The former can be further split into subcomponents, as it has to provide several functionality like dealing with the signaling plane of the conferencing scenario, handling the management of conferences (e.g., creating, modifying, deleting them), and in general taking care of all the business logic, which includes policies, related to the scenario. These policies include Floor Control, which implies that the AS will have to also manage access rights to shared resources in our conferencing framework. The media streams are manipulated and provided by the Media Server. It has to provide the resources, by offering functionality like mixing of incoming media streams (in our case, audio and video streams) and media stream processing (e.g., audio transcoding, media analysis). Finally, considering the XCON framework is conceived to be agnostic with respect to the signaling protocol used to access the service, specific elements are needed as gateways towards other technologies. For instance, a Gateway is needed in order to guarantee the interworking with the Public Switched Telephone Network (PSTN).

2.3 Implementation

As already introduced in Chapter 1, the XCON framework defines a suite of conferencing protocols, which are meant as complementary to the call signaling protocols, for building advanced conferencing applications and achieving

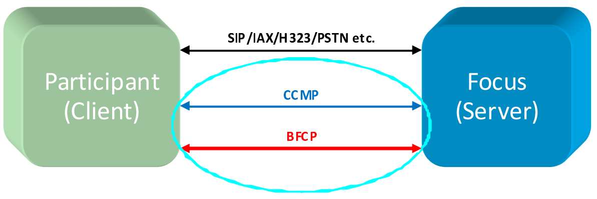


Figure 2.1: New protocols implemented

complex scenarios. These protocols aim at providing means to manage conferences in all their facets.

The realization of an XCON-compliant architecture led us to work both on the client and on the server side, with special focus on all the communication protocols between them and their scenarios of interaction. The client side work included the implementation of both roles envisaged in the architecture, namely the simple participant and the chair. On the server side, we implemented the roles of the *Focus*, as defined in [5], of the Floor Control Server, and of the Media Server. To make the client and server sides interact with each other, we implemented all the envisaged protocols (see Fig. 2.1), specifically BFCP and CCMP. The interaction between the Focus and the Media Server led us to design an additional dedicated protocol for the remote control of the media processing functionality. More details upon this feature will be provided in the following.

As to BFCP, it has been implemented as a dynamic library, which has then been integrated into both client and server entities of the architecture. All the media management, manipulation and delivery have been bound to an event-driven mechanism, according to the directives coming from the Floor Control Server. The CCMP protocol, as it is currently specified in [6], has been implemented and integrated as well, in order to allow clients to dynamically manage conferences creation as well as conferences information.

2.3.1 Server side components

On the server side, we adopted Asterisk[1], a popular open source PBX which is constantly growing in popularity. The modular architecture behind Asterisk design allows it to be quite easily modified and enhanced, upon necessity. Specifically, we added to Asterisk the following new functionality:

- XCON-related identifiers, needed to manage conferences;
- Floor Control Server (FCS), by means of a dynamic library implementing the server-side behavior and policies of the BFCP;
- CCMP Server, the server side component implementing the conference scheduling and management protocol;
- Video Mixer Client, the client side of the protocol implementing the interaction with the remote Video Mixer;
- Notification Service, to enable asynchronous events interception and triggering.

Most of these components have been realized as extensions to a conferencing facility already available as a module in Asterisk, called *MeetMe*. This facility acts as a set of configurable virtual "rooms" for channels that are attached to it, thus allowing users to access conferences by simply calling a predefined phone number, associated with a standard extension of Asterisk's dial-plan, independently from the clients signaling protocol. The addition of the above mentioned functionality allowed us to realize a fully-driven XCON-compliant focus.

The addition of the CCMP component to the "vanilla" MeetMe module allows for dynamic conference management in a user-friendly fashion: in fact, through this component clients are made able to dynamically (i.e., both in an active way, as in scheduling, and in a passive way, as in retrieving information) manipulate the conference objects and instances. Considering

[1]See `http://www.asterisk.org`

the dynamic nature of the framework with respect to policies, settings and scheduled conferences, all the required changes in the dial-plan, as well as dynamic reloading upon necessity, have been accomplished by adding the related functionality to the extended MeetMe module.

For what concerns BFCP, we had to implement the entire protocol, as well as its behavior which includes queues and state machines, from scratch. In order to achieve this, BFCP has been realized as a dynamic library, which is loaded at run time by the Asterisk server and comes into play whenever a resource is to be moderated. In fact, Asterisk, as the entity in charge of the business logic, also acts as the Floor Control Server of the architecture (see Fig. 2.2). The FCS functionality is involved every time a request is generated from a participant, asking for the right to access a specific resource (e.g., audio or video). As suggested by the picture, the FCS itself may or may not take any decision about incoming requests. In fact, while automated policies may be involved to take care of floor control in a more straightforward approach (e.g., to always accept or refuse incoming requests according to predefined policies), if they are not specified the FCS rather forwards floor requests to the designated floor chair, who is in charge of taking a decision that is accordingly notified to all the interested parties. As a transport method for BFCP messages, support for both TCP/BFCP and TCP/TLS/BFCP has been implemented. Besides, since conference-aware participants need to know all the BFCP-related information of a conference in order to take advantage of the BFCP functionality, the focus needs means to provide her/him with such details. Apart from any out-of-band mechanism that could be exploited, the IETF has standardized a way [9] to encapsulate this information within the context of an SDP (Session Description Protocol) offer/answer. This functionality has been implemented as well in the module.

We implemented a Notification Service by exploiting both existing solutions and customized modules. Besides reusing the already available Asterisk Manager Interface (which however only allows active notifications to passive

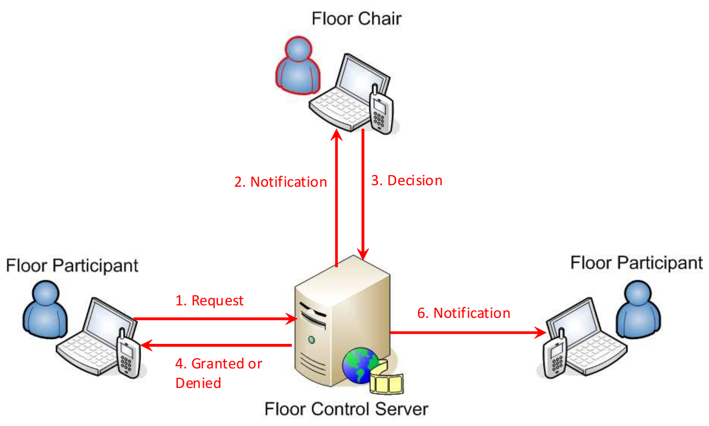


Figure 2.2: The BFCP protocol in action

listeners), we implemented a brand new protocol, which we called *Dispatcher*. This protocol plays a central role when dealing with components distribution in order to improve the scalability of the centralized conferencing framework, as will be explained in Chapter 4.

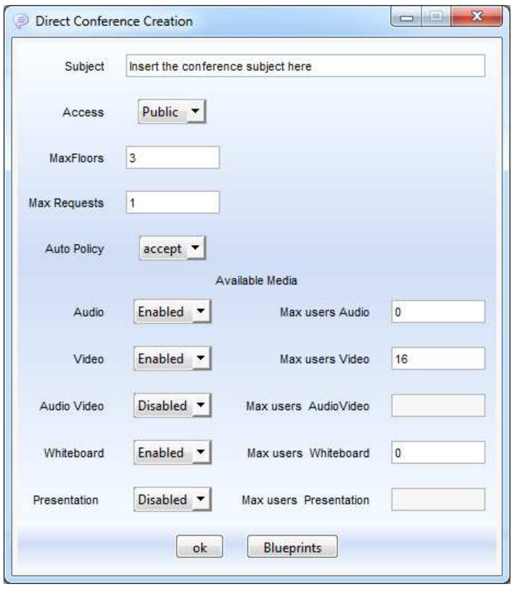
Finally, the existing MeetMe conferencing module provided by Asterisk, which was the basis of our work, only supported audio natively. This obviously was a huge limitation in our framework, both for the user experience and for protocol research interest. In fact, having the possibility to involve moderation on different resources (i.e., not just on audio) provides us with more complex scenarios to deal with. Starting from these considerations, we first paved the way for a video support in the module by adding a basic video-switching functionality. The idea was basically to only allow one participant at a time to contribute to the video feed in a conference: this contribution would then be sent (or better, “switched”) to all the other participants in the conference. This new functionality allowed us to start dealing with a video floor, thus introducing additional complexity in the BFCP interactions and offering interesting research ideas: in fact, the exclusive access to the video resource implied a strong role for the moderation protocol. However, a simple BFCP-moderated video-switching still could not satisfy us for many reasons. Apart from the already mentioned user experience, which could surely benefit from approaches like grid-based video layouts, video-switching, as the name suggests, is a simple blind forwarding of frames coming from a source to one or several destinations. This means that it is in no way concerned with content adaption, which might instead be needed when a conference involves participants making use of heterogeneous applications, devices and/or codecs. The most obvious example is two participants making use of different video codecs (e.g., H.261 and H.263): a blind forwarding would prevent both participants from watching the peer's contribution, if available. This led us to study the possibility of designing an ad-hoc media server which would act as a video multiplexer, for complex layouts involving more sources, and transcoder, to deal with video streams with different encodings and resolutions. To achieve this goal, we designed and implemented a custom video mixer, called *Confiance VideoMixer*. Considering our will to adhere to the separation of responsibilities principle in order to foster scalability, we chose this videomixer to be a remotely controllable media server. In this way, the conferencing module would only have to deal with the application logic (e.g., attaching participants to a mixed stream, specifying mix layouts, and so on), while the videomixer would process and manipulate the video streams according to directives sent by the module. This approach is the same as the one currently fostered by the MEDIACTRL (Media Server Control) Working Group of the IETF, and will be thoroughly examined in Chapter 3. The MeetMe application and the external VideoMixer communicate through a dedicated channel, through which the participants' video streams are controlled. In the current implementation, the protocol allows for the per-user and per-conference customization of several aspects of the video processing, as layouts, transcoding, as well as the optional ability for participants to watch their own contribution in the mix they receive. All the directives the controller (in this case the conferencing module) sends to the videomixer are event-driven, and they make part of its application logic. Whenever a video-enabled participant joins a conference, its stream is redirected to the videomixer. By properly correlating the participant's identifiers associated with its own instances in the controller and in the videomixer, the controller is then able to command different actions on the stream. BFCP moderation is one of the above mentioned events that can result in an action being requested by the controller: the video floor being granted to a participant would have the controller request the related participant's video stream to be included in the overall mix, just as the same floor being denied or revoked would result in an opposite request.

While the CCMP- and BFCP-enabled Asterisk component, empowered with the Confiance Videomixer, provided us with the ability to handle both media streams and signaling, moderation and conference management protocols, we still needed the ability to handle a further protocol, the *eXtensible Messaging and Presence Protocol* (XMPP) [32], we chose as both the Instant Messaging protocol and an out-of-band signaling mechanism of the framework. For such purpose, we chose to make use of a popular XMPP server, called *Openfire*[2], which we placed side by side with Asterisk to realize the logically centralized XCON focus. Openfire, as we will see in Chapter 4, becomes of paramount importance when moving towards a distributed conferencing architecture.

2.3.2 Client side components

On the client side, we adopted an existing open source instant messaging client, called *Spark*[3], as the basis for our work. Spark is a piece of software written in Java, which implements the XMPP protocol to realize instant messaging scenarios. It is natively conceived to interact with the aforementioned Openfire server, and is easily extensible by developing custom plugins which implement new functionality. In our case, we introduced the support for the SIP/SDP, RTP, BFCP and CCMP protocols, besides some additional functionality that will be briefly described in Section 2.4. New graphical widgets have been realized as well, in order to enable user-friendly support for SIP-

[2]See `http://www.igniterealtime.org/projects/openfire/`
[3]See `http://www.igniterealtime.org/projects/spark/`



Direct Conference Creation

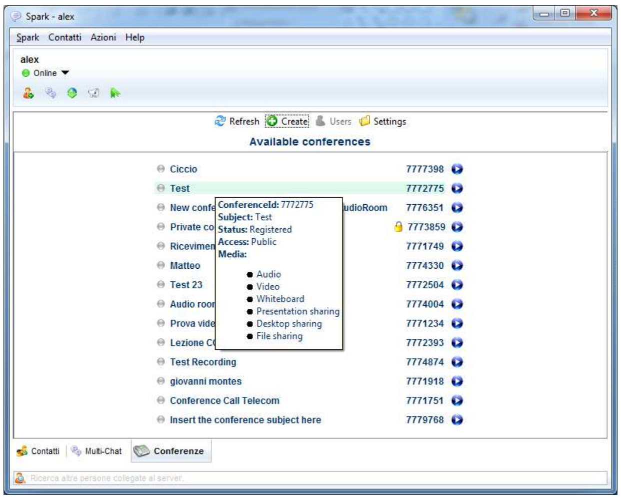
Subject	Insert the conference subject here		
Access	Public		
MaxFloors	3		
Max Requests	1		
Auto Policy	accept		
	Available Media		
Audio	Enabled	Max users Audio	0
Video	Enabled	Max users Video	16
Audio Video	Disabled	Max users AudioVideo	
Whiteboard	Enabled	Max users Whiteboard	0
Presentation	Disabled	Max users Presentation	

Figure 2.3: Conference creation

and BFCP-related settings and to allow users to take advantage of the functionality related to both conference scheduling and BFCP. As to conference scheduling, Fig. 2.3 shows the widget by means of which it is possible to dynamically create new conferences by exploiting the CCMP protocol. Fig. 2.4, instead, shows the list of scheduled conferences, retrieved by means of CCMP as well.

Additionally, we implemented the client side BFCP behavior as a Java library. An ad-hoc panel, associated with such library, has been introduced in order to enable users to:

- Send BFCP messages to the BFCP server;
- Interactively build BFCP floor requests in a user-friendly fashion, either in *participant* or in *chair* (i.e., with enhanced floor management functionality) mode;
- Keep an up-to-date log of all the BFCP messages exchanged with the server (and optionally show each such message in further detail by simply clicking on the related entry in the history widget).



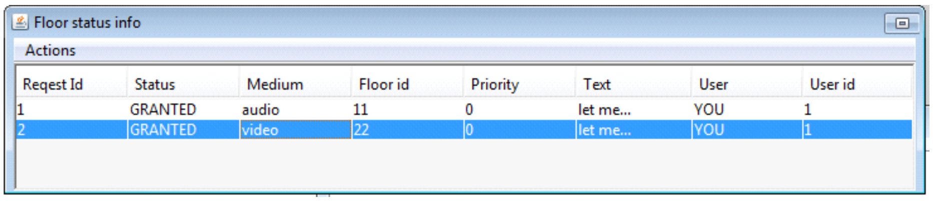
Available conferences

Ciccio	7777398
Test	7772775
New confe… udioRoom	7776351
Private co…	7773859
Ricevimen…	7771749
Matteo	7774330
Test 23	7772504
Audio roo…	7774004
Prova vide…	7771234
Lezione C…	7772393
Test Recording	7774874
giovanni montes	7771918
Conference Call Telecom	7771751
Insert the conference subject here	7779768

ConferenceId: 7772775
Subject: Test
Status: Registered
Access: Public
Media:

- Audio
- Video
- Whiteboard
- Presentation sharing
- Desktop sharing
- File sharing

Figure 2.4: List of scheduled conferences



Reqest Id	Status	Medium	Floor id	Priority	Text	User	User id
1	GRANTED	audio	11	0	let me...	YOU	1
2	GRANTED	video	22	0	let me...	YOU	1

Figure 2.5: Moderation panel

With respect to the role of the chair, we added ad-hoc interfaces in order to enable potential moderators to either manage floor requests issued by conference participants (an example of such interfaces is shown in Figure 2.5), or build so-called *third-party* floor requests, i.e., requests generated by the chair on behalf of a different participant. It is worth noting that such functionality is particularly interesting since it enables the chair to allow conference-unaware participants to take part to an XCON-enabled conference.

Finally, as to the VoIP functionality of the client, we adopted and extended an open source SIP stack called *MjSip*[4]. To take advantage of the already mentioned negotiation of BFCP information within the context of the SDP offer/answer, we added the support for the encapsulation of BFCP information in SDP bodies. In this way, the BFCP is automatically ex-

[4]See `http://www.mjsip.org`

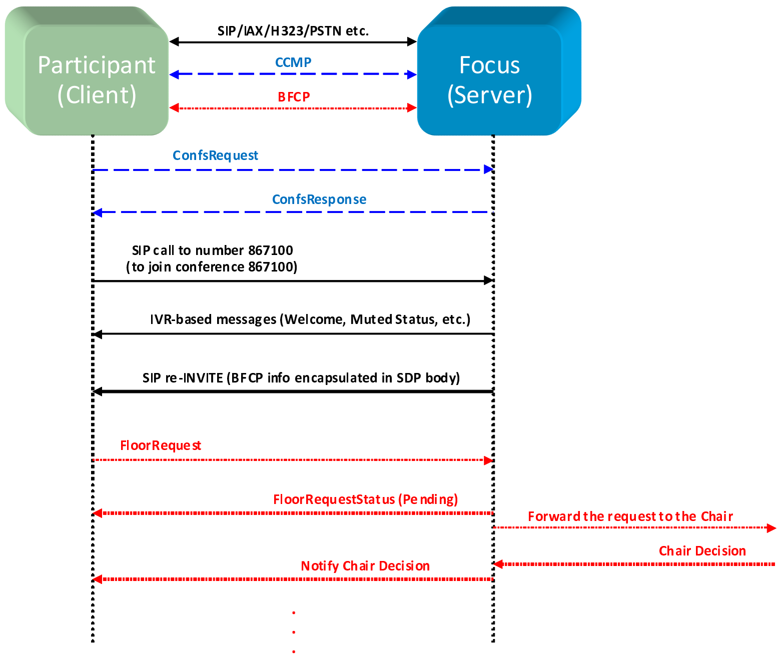

Participant (Client)
Focus (Server)
SIP/IAX/H323/PSTN etc.
CCMP
BFCP
ConfsRequest
ConfsResponse
SIP call to number 867100 (to join conference 867100)
IVR-based messages (Welcome, Muted Status, etc.)
SIP re-INVITE (BFCP info encapsulated in SDP body)
FloorRequest
FloorRequestStatus (Pending)
Forward the request to the Chair
Chair Decision
Notify Chair Decision

Figure 2.6: An example of signaling between client and server

ploited whenever a SIP INVITE (or re-INVITE, in case the negotiation is involved in a subsequent moment) contains BFCP-related identifiers. Besides, the appropriate transport method for the BFCP communication with the FCS (i.e., TCP/BFCP or TCP/TLS/BFCP) is automatically chosen and exploited with respect to this SDP negotiation.

2.3.3 An example of client-server interaction

To provide the reader with a more detailed overview of the way the client-server interaction involves the introduced protocols, this subsection is devoted to presenting an example regarding a typical use case scenario. To ease the understanding of the sequence diagram depicted in Fig. 2.6, each protocol is represented with a different line style:

1. A participant (client in the scenario) contacts the Focus (the server), through the CCMP protocol (dashed line), to ask for the list of currently active conferences, thus sending a *ConfsRequest* message request

with *Active* as argument;

2. The Focus processes the request and sends back to the participant (still through the CCMP protocol) a *ConfsResponse* message, containing the list of all active conferences;

3. The participant reads the list and decides to join the active conference identified by the number 8671000: to join the conference, she/he calls the conference number – as if it were a standard phone number – using SIP (solid line) as the call signaling protocol, thus placing a call to the SIP URI 8671000@*Focus* (where Focus is the SIP domain, in this case the IP address of the Asterisk server);

4. The Focus receives the call and, according to the specified dialplan rules, routes it to the XCON-enabled MeetMe instance managing the conference with the same call number;

5. The XCON-enabled MeetMe instance managing the conference, through IVR (*Interactive Voice Response*), plays back a series of pre-recorded voice messages to welcome the new user. It also warns the client about the fact that she/he is initially muted in the conference;

6. All the relevant BFCP information is encapsulated in an SDP body, and then sent back to the new user by means of a SIP re-INVITE;

7. Once the client receives the re-INVITE and becomes aware of the needed BFCP set of data, she/he, using the BFCP (dotted line), decides to make a *FloorRequest* to ask the Focus for the permission to talk;

8. The Focus, as Floor Control Server, answers the client by sending back a *FloorRequestStatus* BFCP message notifying that the request is currently pending. At the same time, the Floor Control Server forwards the message to the chair of the requested floor as well, to ask him to take a decision about the request.

From this point on, the BFCP transaction proceeds exactly as described before (see Fig. 2.2). Once the chair grants the floor, the client is un-muted and thus given the permission to talk until the floor is not willingly released by the client herself/himself or revoked by the chair. Since a floor is a logical object, all BFCP transactions will proceed in the same way, independently from the set of resources (be it audio or video, in the case of our platform) the related floor(s) could be associated with. In case the floor request involved a manipulation of a video request, a subsequent interaction between the conferencing module and the remote videomixer would take place through the dedicated channel.

2.4 Additional functionality

In this section we provide a brief overview of the additional functionality we implemented in the Meetecho conferencing system. Most of such functionality have been realized by exploiting open source software programs, which have been extended and modified to meet our requirements.

2.4.1 Whiteboarding and polling

The first additional feature we introduced in the platform is a whiteboarding and polling tool, which allows users participating in a conferencing session to share one or more common drawing areas and to make polls and voting. We started from the open source tool *jSummit*[5], making a lot of changes to it in order to migrate from a peer to peer approach to a more suitable to us client-server paradigm. The server side of the protocol is always hosted by the XCON focus, while the client side has been introduced in the aforementioned plugin for the Spark client.

[5]See `http://jsummit.sourceforge.net`

2.4.2 Slides sharing

A feature typically requested in conferencing systems is presentation sharing. This feature basically allows one of the participants, the presenter, to share a presentation with the other participants, and to discuss its slides accordingly. A presentation might be shared in several different formats, like Microsoft PowerPoint, Adobe PDF, Open Document Format. We dealt with such heterogeneity of formats by considering every presentation as a simple slideshow of static images. In fact, whenever a presentation is shared in a conference, it is converted in background by the server to a series of images made available on a web server together with metadata information. Besides, whenever the presenter triggers a slide change, such event is notified to the other participants by means of XMPP: information about the new slide is provided in the notification (the slide number, the HTTP URL where the slide can be retrieved, etc.) making it easy for the other participants to passively attend the presentation.

2.4.3 Desktop sharing

Another functionality usually offered by conferencing systems is desktop sharing. We introduced such feature in Meetecho by integrating a Java-based open source tool called *JRDesktop*[6]. It leverages the Java Remote Method Invocation (RMI) mechanism in order to let clients, called *Viewers*, get screens updates from the *Sharing Server*. Such "pull" paradigm did not seem to us as the best solution for two reasons: (i) the Sharing Server has to handle as many connection as the actual number of viewers connected, meaning that bandwidth and resource consumption become critical and might affect other functionality like audio/video streaming; (ii) a "pull" approach is not suitable when the sharing server is within a private network environment, with a Network Address Translator (NAT) being its interface with the public Internet. In order to address such issues, we modified the JRDesktop software by introducing a new role, the *Reflector*, which acts as a proxy be-

[6]See `http://jrdesktop.sourceforge.net/`

tween the sharing server and the viewers by forwarding the screens updates to all interested parties. The connection with the reflector is the only one the sharing server has to handle, thus saving bandwidth and CPU cycles. Furthermore, such connection is initiated by the Sharing Server, subverting the usual client-server paradigm, in order not to be sensitive to any possible NAT/Firewall. We realized such behavior taking inspiration from the FTP protocol [22] and its "passive" operation mode: after a signaling phase performed by means of XMPP, during which IP addresses and port numbers are exchanged, two custom RMI socket factories we implemented are invoked on both sides to establish the channel.

Finally, the remote control functionality was already envisaged by the JRDesktop software, meaning that every viewer might take the control of the remote desktop. We just added moderation to such resource by means of BFCP.

2.4.4 Session recording

While online and real-time collaboration already has a strong value per se, the ability of recording a conferencing session and playing it out ex-post would definitely provide added value to any conferencing environment. In fact, a recorded conferencing session can be seen as an important media asset, which can play an important role in several scenarios, like e-learning, minutes and so on. Of course, recording a multimedia conferencing session does present many challenges we had to face, considering the number of media that may be involved asynchronously. In fact, a multimedia conferencing session may involve several different media at the same time. Besides, those media may come and go asynchronously. This is especially true in our Meetecho conferencing platform, which allows for the dynamic addition of heterogeneous media to a conference, like audio, video, instant messaging, whiteboards, shared presentations and so on. As a consequence, it is quite obvious that, in order to achieve a proper recording of a conferencing session, just dumping the protocol contents associated with each media and storing them is likely not enough. At least additional timing information is needed, in order to be able to contextualize each involved medium in one or more time frames, and allow for inter-media synchronization. The same can be said for relevant events that may occur during the lifetime of a conference. This is exactly the approach we took towards the recording of conferencing sessions. For each medium, we devised a way to record and store the relevant information, together with related metadata. A post processing phase may subsequently be involved in order to take care of the fusion of metadata information. In order for these recorded assets to be actually valuable to an interested user, they need to be made available in a proper way that takes into account the relationship among the original media, as well as their synchronization. This led us to look for a standard way to correlate such heterogeneous media between each other, while also taking into account timing information expressing when a medium appears in a conference, or when any relevant event happens. The solution we came up with was the exploitation of a well known and established standard specification, the *Synchronized Multimedia Integration Language* (SMIL) [34]. SMIL is basically an XML-based markup language which defines a standard for describing presentations involving heterogeneous media. As such, it is the perfect candidate for a successful playout of a recorded multimedia session. For the sake of conciseness, we do not provide any further detail on such topic. The interested reader may refer to [3].

Chapter 3

Towards scalable conferencing: the MEDIACTRL approach

3.1 Introduction

The more users ask for value added applications, the more an evolution of the obsolete network infrastructure and architecture is needed. To achieve the goal of granting a transparent (with respect to both devices and access networks) and better fruition of both contents and services, major efforts are being directed in separation of concerns among network components and heterogeneity of access. The separation of responsibilities is not a new proposal when thinking about multimedia capabilities. Several approaches have been presented in the past to cope with such an issue, some of them even within a standardization context. The first approach that comes to mind is the H.248 protocol [13], also known as MeGaCo as it was called when standardized by the IETF. This protocol, based on XML payloads, allowed applications to invoke media services from a remote Media Server by means of a low-level API. Despite a widespread deployment of implementations supporting this specification, its low-level approach has recently moved the interested parties into researching an alternative way of dealing with media services. The preferred path of research almost suddenly became a SIP-based approach, SIP being the de-facto standard for IP-based multimedia applications and services. This led to several proposals, some already standardized and some still being specified. Such an abundance of proposals obviously resulted in a potential issue for implementors, considering a single standard reference protocol to be used in conjunction with SIP was still missing. As to the aforementioned matter, in this chapter we focus on the activities currently being carried out within the IETF by the Media Server Control Working Group (MEDIACTRL).

The MEDIACTRL approach proved very useful to us, as it has been adopted when designing the interaction between the XCON focus and the VideoMixer component of the Meetecho conferencing system (see Chapter 2). In Section 3.5 we will touch on how MEDIACTRL and its separation of responsibilities principle let us make the first step towards a scalable multimedia conferencing system.

3.2 Media Server Control

The approach taken by the Internet Engineering Task Force is separating the application logic from the media processing. After investigating the needed requirements, the MEDIACTRL Working Group was opened, which specifies the Media Server (MS) as a centralized component that an Application Server (AS) can interact with by means of a dedicated protocol in order to implement multimedia applications. Hence, the MEDIACTRL WG aims at specifying an architectural framework to properly cope with the separation of concerns between Application Servers (ASs) and Media Servers (MSs) in a standardized way. As such, the MEDIACTRL architecture envisages several topologies of interaction between AS and MS, the most general one being an m:n topology. Nevertheless, our focus is on a 1 : 1 interaction (see Fig. 3.1).

The current specification of the framework [19, 7] envisages a modular approach when looking at the functionality to provide. This means that, inside the same MS, different inner components take care of orthogonal processing that may be required. To achieve this, the framework currently specifies a general invocation mechanism with opaque payloads, whereas such payloads can be directed to the proper component according to a header in the

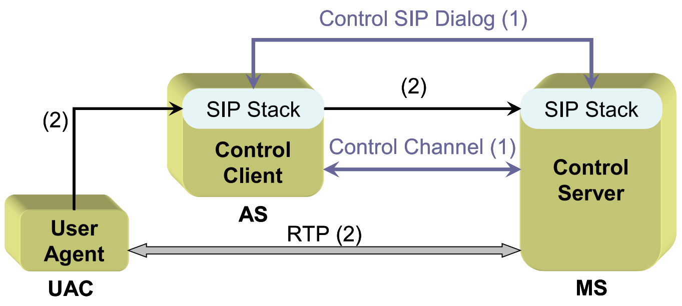


Figure 3.1: MEDIACTRL architecture

request itself. This way, new components providing additional media capability can be added at any time without affecting the specification of the core framework. Such components are called Control Packages; so far two different packages have been proposed: a package providing basic Interactive Voice Response functionality [18], and a package for managing mixers for media conferences and connections [17].

When looking at the protocol itself, the interaction between an AS and a MS relies on a so-called Control Channel. This channel is where MEDIACTRL-related messages flow between the AS (the client side) and the MS (the server side). An AS would instruct a MS into a specific media operation by placing a request on this channel, and the MS would reply to such request, as well as notify events, through the same channel. Of course, such a channel can only be set up as the result of a transport-level connection. This implies that either the AS or the MS must previously know the transport address of the other party in order to set up the connection. To allow this, a COMEDIA-based approach [35] has been defined: the AS and MS make use of a SIP dialog to negotiate and set up the Control Channel.

Once this channel has been opened, a way to have the AS and MS authenticate each other is of course needed. This is needed in order to make sure that the client opening the control channel with the server is actually the same that initiated the SIP dialog in the first place. Such an authentication is accomplished by means of a dedicated Control Channel method called `SYNCH`: the newly connected AS has to send a properly constructed

`SYNCH` message to the MS right after the connection has been opened, otherwise the connection is torn down.

In the following subsections, we provide further details about the AS and MS roles which, in our Meetecho conferencing platform, are played by the XCON focus element and by the Confiance VideoMixer component, respectively.

3.2.1 Application Server: the brain

The AS component plays a role which is of paramount importance inside MEDIACTRL. It is in charge of appropriately controlling a MS in order to provide advanced services to end-users, and as such it is where all the application logic related to the services resides. To make a very simple example, it can be seen as the *brain* in the architecture, the entity making decisions and controlling all the actions accordingly.

The establishment of a Control Channel has already been introduced. This establishment is a prerequisite to any further interaction between the AS and the MS.

For what concerns end-users, instead, being the AS a frontend to them, it is also in charge of terminating the signaling, in this case SIP. Considering the media functionality is actually provided by the MS and not by the AS itself, a way to have the AS transparently attach the users' media connections to the MS is needed: these media connections would then need to be properly manipulated to implement the service itself. In order to achieve this result, the specification envisages the use of a 3rd Party Call Control (3PCC) mechanism: the AS terminates the SIP signaling with the end-users and forwards their requests to the MS in order to have all media connections negotiated accordingly. These media connections are subsequently referenced by both the AS and the MS when needed (e.g., when the AS wants the MS to play an announcement on a specific user's connection).

3.2.2 Media Server: the arm

Just as the AS is responsible for all the business logic, the MS is conceived to take care of every facet of the media processing and delivery. Its operations are realized according to the directives coming from the controlling AS. To recall the previously presented example, while the AS is the brain, the MS is the *arm*.

Such a distinction in roles makes it clear that the MS is supposed to be directly responsible for all the media on a low level. This means that, besides acting as the termination point for media connections with end users (whereas signaling is terminated by the AS), the MS is also responsible for manipulating these media connections according to incoming directives, policies and previous negotiations. Examples of operations a MS is supposed to be able to achieve include:

- Mixing, transcoding and adaptation of media streams;
- Low-level manipulation of media streams (e.g., gain levels, video layouts, and so on);
- Playing and recording of media streams from and to both local and network environments;
- Storing and retrieving of external references of any kind (e.g., media streams, VoiceXML and SRGS directives, and so on);
- Tone and DTMF detection;
- Text-To-Speech and Speech Recognition.

Whereas a limited set of such operations is implicitly accomplished by the MS as a consequence of the initial SIP negotiations that make it aware of end users (e.g., user A only supports GSM audio, while user B also supports H.263 video as long as the bit rate is limited to 10kbps), the most relevant tasks for the MS come from the requests made by authorized ASs. For instance, within the application logic of a conferencing scenario an AS may first attach an end user to the MS in order to have them negotiate the available media between each other, and subsequently instruct the MS into playing an announcement (e.g., "Digit the PIN") followed by a DTMF collection (to have the user digit the conference pin number), and then into joining the user into the conference mix itself.

All these requests, together with the related responses and event notifications, flow through the already discussed control channel just as the previously described `SYNCH` transaction does.

3.3 An open-source implementation

The standardization work on the MEDIACTRL architecture is almost completed. In order to help researchers and developers understand the architecture and possibly dig out the flaws that may be hidden inside its specification, we provided an open source implementation of the specified architecture, following the well known IETF motto "rough consensus, running code" Our implementation has been developed on both the server and client sides, and currently includes the core framework, the new MEDIACTRL protocol and some of the control packages that can be employed in order to implement custom media manipulation. All the relevant details upon the implementation choices will be presented in the following subsections.

It is worth noting that our prototype implementation also paved the way for a call flow document [4] we are carrying on in the MEDIACTRL Working Group. In fact, having a real world implementation of the protocol easily allowed us to reproduce popular use case scenarios involving media by means of the MEDIACTRL architecture, and consequently first-hand protocol interactions between AS and MS.

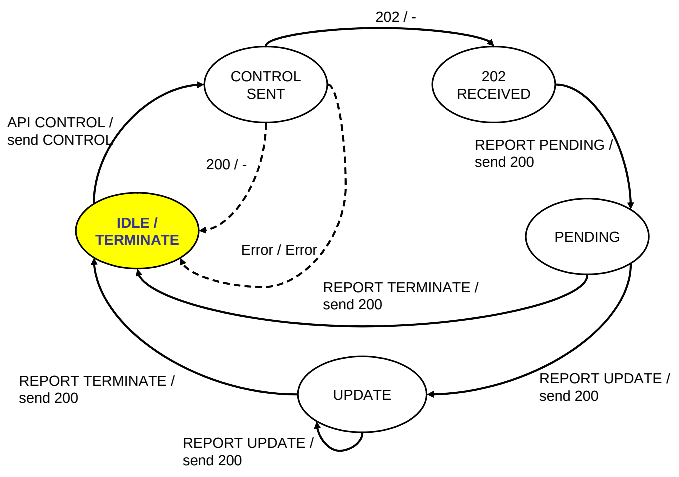


Figure 3.2: The Application Server perspective in MEDIACTRL

3.3.1 Application Server: Asterisk

Starting from the client side, we had to take into account what was required from a MEDIACTRL-enabled Application Server besides the business logic itself. Considering that the specification clearly points out that such an AS must support SIP, both for setting up the control channel with a MS by means of the previously introduced COMEDIA negotiation and for attaching User Agents to the MS through a 3PCC mechanism, once again the Asterisk server introduced in Chapter 2 was perfectly suitable for us, as it provided all the needed functionality.

The AS is in charge of appropriately controlling a Media Server in order to provide advanced services to end-users. Fig. 3.2 gives a simplified view of the protocol behavior of an AS interacting with a Media Server.

3.3.2 Media Server: Confiance VideoMixer

Coming to the MS itself, instead, the requirements were quite different than the one we identified for the client side. In fact, as explained in the previous section, a MEDIACTRL-enabled MS envisages low level media processing and manipulation besides custom media delivery by means of RTP.

This obviously suggested us that such an implementation would have

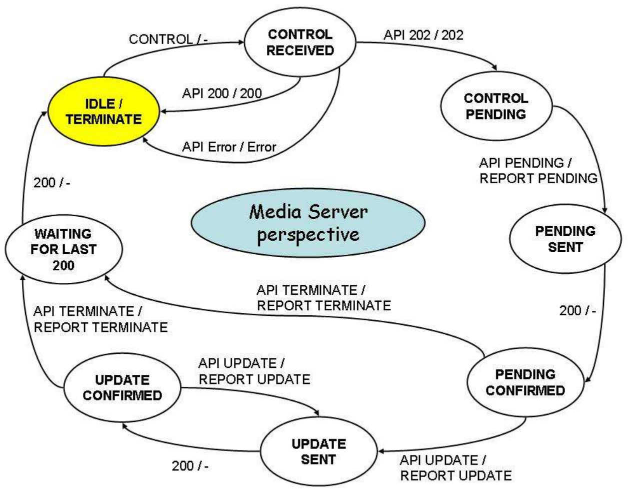


Figure 3.3: Media Server SCFW State Diagram

strict real-time requirements, which led us into choosing C/C++ as the programming language to use to implement it.

However, the preliminary step to any actual implementation of the framework was indeed the investigation of the possible protocol states. This led us into specifying a state diagram for the MS as depicted in Fig. 3.3, which paved the ground for the implementation work itself.

While most of the code was written from scratch, we also decided to avoid the *not invented here* syndrome and thus reuse as many existing open source components as possible in the implementation.

The most important protocols to deal with were of course SIP and SDP. We needed a powerful and flexible SIP stack, capable to provide us with means to properly handle the 3PCC mechanism, as well as means to easily extend the standard SIP protocol with respect to both headers and bodies, in order to cope with the additional functionality envisaged in the MEDIACTRL specification (e.g., the COMEDIA negotiation and the ctrl-package SDP attribute). Thus, we chose the C++ *reSIProcate*[1] library as our SIP stack, which provided us with powerful means to deal with SIP behavior. This library is indeed well known in the open source community, and has

[1]See `http://www.resiprocate.org`

many active IETF participants among its contributors. Besides, it has also been successfully used in many commercial projects as well, including two widespread SIP softphones, namely X-Lite and Eyebeam.

When coming to RTP, instead, we decided to make use of the open source C library *oRTP*[2].

Considering all the currently specified packages make a heavy use of XML for the framework messages bodies, we also had a strong need for a reliable XML parser. Our choice fell on the widespread *Expat*[3] library, a very well known lightweight open source C component. The actual parsing of the XML contents, anyway, was left to an additional component called *Boost::regex*.

Nevertheless, the list of protocols and meta-languages to handle did not end here. In fact some packages currently require the support for retrieval of external references. It is the example, for instance, of the Basic IVR package, which allows in its specification to externally refer resources to be used within dialogs, as remote media streams to play out in announcements, stored XML files containing the actual directives for the package, SRGS grammar bodies and so on. In order to appropriately satisfy this requirement, we made use of another well known open source component called *libcurl*[4]. This library allows for an easy retrieval of external files of any kind by supporting a wide range of protocols, including HTTP, FTP and many others.

Of course, all these libraries only allowed us to handle the protocols needed by the MEDIACTRL framework. Nevertheless, the framework would not be complete without an actual media support with respect to codecs, in order to properly implement the needed transcoding functionality. This led us to make use of additional open source components, the most important being the very well known *ffmpeg*[5] piece of software. This software in fact provides an easy API, called *libavcodec*, to deal with decoding and encoding of media streams, and proved very useful especially when implementing our

[2]See `http://www.linphone.org`

[3]See `http://expat.sourceforge.net`

[4]See `http://curl.haxx.se`

[5]See `http://ffmpeg.mplayerhq.hu`

video support for the framework.

All the aforementioned libraries were then integrated in the code we implemented from scratch ourselves, which will be briefly described in the following lines with respect to the requirements we met.

Particular attention was required by the way media connections are conceived in the specifications. In fact, such media connections can be addressed at different levels of granularity, according to whether the specific media label is included or not in the connection identifier. Hence, we specified wrapper classes and related callbacks in order to properly drive the flow of media accordingly.

For what concerns the MS itself, it was conceived to be modular with respect to both codecs (audio and video) and control packages, for which we designed a dedicated API. This allowed us to separate the design of the core framework itself from the realization of proof of concept control packages and codecs used to test the MS functionality. Our current implementation offers good support to both audio (we support G.711 and GSM codecs) and video (H.263 and H.264 codecs). Regarding the control packages, we focused on two of the previously introduced ones, specifically Basic IVR and Conferencing. This allowed us to reproduce many real-world scenarios by having the AS properly orchestrate requests to the MS.

3.4 Use Case Scenarios

The presented implementation efforts allowed for the testing of proper real-world use case scenarios involving media processing and delivery. Many of these scenarios have been included in the already mentioned call flow document [4], where snapshots of the full protocol interaction between AS and MS for each of them are provided.

Having implemented two orthogonal packages with respect to the provided functionality, we were able to design several heterogeneous scenarios, ranging from simple echo tests to phone calls, conferences and voice-mail applications. Each scenario could be reproduced by properly orchestrating different requests and by correlating the resulting output, just like a state machine. As a reference scenario to describe in this chapter we chose the *Echo Test* one, which basically consists of a UAC directly or indirectly "talking" to itself. The choice is motivated by the fact that, despite being a really simple example, this scenario can be achieved in several different ways, and so proves to be quite useful when it comes to describing its implementation making use of different approaches. Other scenarios can be achieved taking quite similar approaches, by integrating different transactions in different fashions within the application logic.

The upcoming scenario description will focus on the specifically scenario-related interaction between the AS and the MS and the results in the UAC experience. This implies that the example assumes that a Control Channel has already been correctly established and SYNCHed between the reference AS and MS as described in the previous sections. Similarly, the 3PCC session among the AS, the MS and the interested UAC is also assumed to have already happened.

Once all the preliminary steps have taken place, the actual scenario can be reproduced and analyzed. We will herein provide the description of two different ways an Echo Test scenario can be achieved, namely:

1. A Direct Echo Test approach, where the UAC directly talks to itself;
2. A Recording-based Echo Test approach, where the UAC indirectly talks to itself.

3.4.1 Direct Echo Test

In the Direct Echo Test approach, the UAC is directly connected to itself. This means that each frame the MS receives from the UAC is sent back to it in real-time.

In the framework this can be achieved by means of the conference control package, which is in charge of the task of joining connections and conferences.

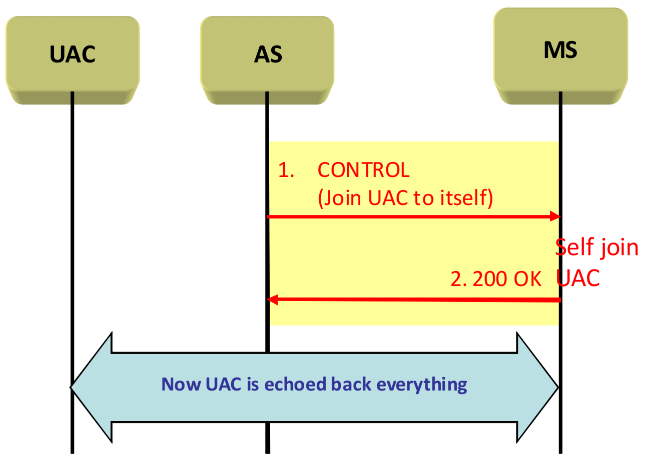

UAC
AS
MS
1. CONTROL (Join UAC to itself)
Self join UAC
2. 200 OK
Now UAC is echoed back everything

Figure 3.4: Self Connection: Framework Transaction

Specifically, the package method the AS has to make use of is called `<join>`, and a sequence diagram of a potential transaction is depicted in Figure 3.4.

All the transaction steps have been numbered to ease the understanding of the subsequent explanation lines:

- The AS requests the joining of the connection to itself by sending a CONTROL request (1), specifically meant for the conferencing control package (`msc-conf-audio/1.0`), to the MS: since the connection must be attached to itself, the `id1` and `id2` attributes are set to the same value, i.e., the `connectionid`;
- The MS, having checked the validity of the request, enforces the join of the connection to itself; this means that all the frames sent by the UAC are echoed back to it; to report the success of the operation, the MS sends a 200 OK (2) in reply to the MS, thus ending the transaction.

The complete transaction, that is the full bodies of the exchanged messages, is provided in the following lines:

1. AS → MS (SCFW CONTROL)

```
SCFW 74b0dc511949 CONTROL
Control-Package: msc-conf-audio/1.0
```

```
Content-Type: text/xml
Content-Length: 87

<?xml version="1.0"?>
<join id1="1536067209~913cd14c" \
      id2="1536067209~913cd14c">
</join>
```

2. AS ← MS (SCFW 200 OK)

```
SCFW 74b0dc511949 200
Content-Type: text/xml
Content-Length: 70

<?xml version="1.0"?>
<response status="200" reason="Join successful"/>
```

Such a transaction is the simplest form of transaction that can occur through the Control Channel. In fact, the reply to the CONTROL message is immediately provided to the AS in a 200 message. This is not always true, since asynchronous events related to the original request may occur and consequently influence the AS state behavior.

3.4.2 Echo Test based on Recording

In the Recording-based Echo Test approach, the UAC is connected to itself in an *indirect* fashion. This means that each frame the MS receives from the UAC is first recorded: then, when the recording process ends, the whole recorded frames are played back to the UAC as an announcement. A well known application making use of this approach is Skype, which envisages three steps: (i) first, an announcement is played to the user agent (e.g., “This is an echo test, talk after the beep for 10 seconds”); (ii) then, a recording of the media sent by the user takes place; (iii) finally, the recording is played back to the user, in order to make it aware of how his media would be perceived by a peer.

In MEDIACTRL, the presented three steps can be reproduced by means of the basic IVR control package, which is in charge of the task of first recording and then playing out the recorded frames. Nevertheless, the whole scenario cannot be accomplished in a single transaction. At least two steps, in fact, have to be made:

1. First, a recording (preceded by an announcement, if requested) must take place;

2. Then, a play-out of the previously recorded media must occur (which again can be preceded by an announcement, if the AS wishes so).

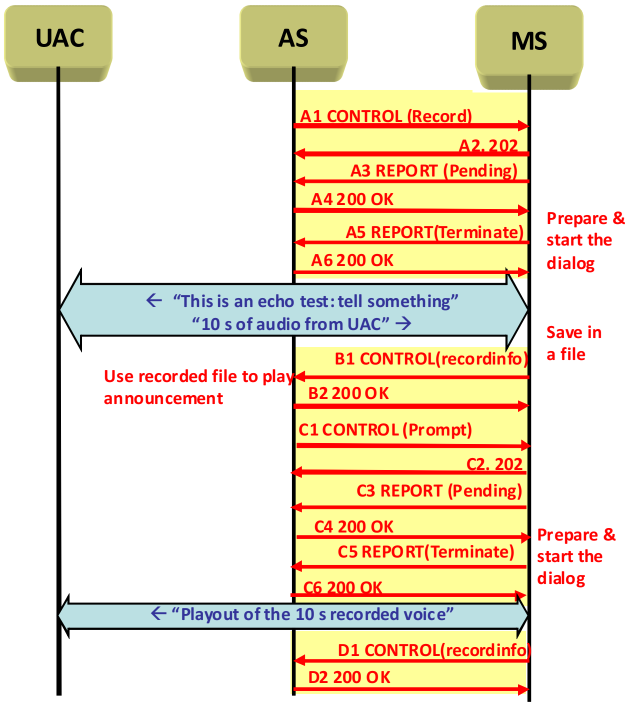


Figure 3.5: Recording-based Echo: Two Framework Transactions

This means that two separate transactions need to be invoked. A sequence diagram of a potential multiple transaction is depicted in Figure 3.5.

Notice how the AS-originated CONTROL transactions are terminated as soon as the requested dialogs start: as specified in [18], the MS makes use of a framework CONTROL message to report the result of the dialog and how it has proceeded. The two transactions (the AS-generated CONTROL request and the MS-generated CONTROL event) are correlated by means of the associated dialog identifier, as it will become clearer from the following lines. As before, all the transaction steps have been numbered to ease up the understanding of the subsequent explanation lines. Besides, the two transactions are distinguished by the preceding letter (A,B=recording, C,D=playout).

- The AS, as a first transaction, invokes a recording on the UAC connection by means of a CONTROL request (A1); the body is for the IVR package (`msc-ivr-basic/1.0`), and requests the start (`dialogstart`) of a new recording context (`<record>`); the recording must be preceded by an announcement (`<prompt>`), must not last longer than 10s (`maxtime`), and cannot be interrupted by a DTMF tone (`dtmfterm=false`);

this has only to be done once (`iterations`), which means that if the recording does not succeed the first time, the transaction must fail; a beep has to be played (`beep`) right before the recording starts, to notify the UAC;

- As seen before, the first responses to the request start flowing: the provisional 202 (A2), the subsequent REPORT pending (A3), and its related ACK (A4) from the AS;
- In the meanwhile, the MS prepares the dialog (e.g. by retrieving the announcement file, for which a HTTP URL is provided, and by checking that the request is well formed) and if all is fine it starts it, while notifying the AS through a new REPORT (A5) with a terminated status: the connection is then passed to the IVR package, which first plays the announcement on the connection, followed by a beep, and then records all the incoming frames to a buffer;
- The AS acknowledges the latest REPORT (A6), thus terminating this transaction, and starts waiting for the result to come;
- Once the recording is over, the MS prepares a notification CONTROL (B1) whose body contains (`<recordinfo>`) the path to the recorded file (in this case, a HTTP URL) which can be used by the AS in case of need;
- The AS concludes this first recording transaction by acknowledging the CONTROL event (B2).

Now that the first transaction has ended, the AS has the 10s recording of the UAC talking. It can let the UAC hear it by having the MS play it as an announcement:

- The AS, as a second transaction, invokes a play-out on the UAC connection by means of a new CONTROL request (C1); the body is once again for the IVR package (`msc-ivr-basic/1.0`), but this time it requests the start (dialogstart) of a new announcement context (`<prompt>`); the file to be played is the one recorded before (`media`), and has only to be played once (`iterations`);
- Again, the usual provisional 202 (C2), the subsequent REPORT pending (C3), and its ACK (C4) from the AS take place;

- In the meanwhile, the MS prepares the new dialog and starts it, notifying the AS about it with a new REPORT (C5) with a terminated status: the connection is then passed to the IVR package, which plays the file on it;
- The AS acknowledges the terminating REPORT (C6), now waiting for the announcement to end;
- Once the play-out is over, the MS sends a CONTROL event (D1) which contains in its body (`<promptinfo>`) information about the just concluded announcement;
- The AS concludes this second and last transaction by acknowledging the CONTROL event (D2).

As in the previous paragraph, the whole SCFW interaction is provided for a more in depth evaluation of the protocol interaction.

A1. AS → MS (SCFW CONTROL, record)

```
SCFW 74b0dc511949 CONTROL
Control-Package: msc-ivr-basic/1.0
Content-Type: text/xml
Content-Length: 354

<?xml version="1.0"?>
<dialogstart connectionid="1536067209~913cd14c">
  <basicivr>
    <prompt iterations="1">
      <media src="http://www.ms.org/prompts/connected.wav" \
         type="audio/wav"/>
    </prompt>
    <record maxtime="10s" dtmfterm="false" beep="true"/>
  </basicivr>
</dialogstart>
```

A2. AS ← MS (SCFW 202)

```
SCFW 74b0dc511949 202
```

A3. AS ← MS (SCFW REPORT pending)

```
SCFW 74b0dc511949 REPORT
Seq: 1
Status: pending
Timeout: 10
```

A4. AS → MS (SCFW 200, ACK to 'REPORT pending')

```
SCFW 74b0dc511949 200
Seq: 1
```

A5. AS ← MS (SCFW REPORT terminate)

```
SCFW 74b0dc511949 REPORT
Seq: 2
Status: terminate
Timeout: 10
Content-Type: text/xml
Content-Length: 88

<?xml version="1.0"?>
<response status="200" \
      reason="Dialog started" dialogid="05ded7b"/>
```

A6. AS → MS (SCFW 200, ACK to 'REPORT terminate')

```
SCFW 74b0dc511949 200
Seq: 2
```

B1. AS ← MS (SCFW CONTROL event)

```
SCFW 4rgth45632d1 CONTROL
Control-Package: msc-ivr-basic/1.0
Content-Type: text/xml
Content-Length: 197

<?xml version="1.0"?>
<event dialogid="05ded7b">
  <dialogexit status="1">
    <recordinfo termmode="maxtime" duration="10000" \
        size="161644" type="audio/wav" \
    recording="http://www.ms.org/recording-05ded7b.wav"/>
  </dialogexit>
</event>
```

B2. AS → MS (SCFW 200, ACK to 'CONTROL event')

```
SCFW 4rgth45632d1 200
```

C1. AS → MS (SCFW CONTROL, play)

```
SCFW 238e1f2946e8 CONTROL
Control-Package: msc-ivr-basic/1.0
Content-Type: text/xml
Content-Length: 319

<?xml version="1.0"?>
<dialogstart connectionid="1536067209~913cd14c">
  <basicivr>
    <prompt iterations="1">
      <media src="http://www.ms.org/recording-05ded7b.wav" \
         type="audio/wav"/>
    </prompt>
  </basicivr>
</dialogstart>
```

C2. AS ← MS (SCFW 202)

```
SCFW 238e1f2946e8 202
```

C3. AS ← MS (SCFW REPORT pending)

```
SCFW 238e1f2946e8 REPORT
Seq: 1
Status: pending
Timeout: 10
```

C4. AS → MS (SCFW 200, ACK to 'REPORT pending')

```
SCFW 238e1f2946e8 200
Seq: 1
```

C5. AS ← MS (SCFW REPORT terminate)

```
SCFW 238e1f2946e8 REPORT
Seq: 2
Status: terminate
Timeout: 10
Content-Type: text/xml
Content-Length: 88

<?xml version="1.0"?>
<response status="200" \
      reason="Dialog started" dialogid="6faf4e0"/>
```

C6. AS → MS (SCFW 200, ACK to 'REPORT terminate')

```
SCFW 238e1f2946e8 200
Seq: 2
```

D1. AS ← MS (SCFW CONTROL event)

```
SCFW g56dhg73g8r5 CONTROL
Control-Package: msc-ivr-basic/1.0
Content-Type: text/xml
Content-Length: 165

<?xml version="1.0"?>
<event dialogid="6faf4e0">
  <dialogexit status="1">
    <promptinfo termmode="completed" \
        duration="10000" iterations="1"/>
  </dialogexit>
</event>
```

D2. AS → MS (SCFW 200, ACK to 'CONTROL event')

```
SCFW g56dhg73g8r5 200
```

3.5 MEDIACTRL in Meetecho

In Chapter 2 we described the current implementation of the Meetecho conferencing system and introduced two main server-side elements: the *Asterisk* server and the *Confiance VideoMixer* component. The former plays the role of the *Focus*, as envisaged by the XCON specifications, while the latter takes care of media streams manipulation and handling. We implemented the interaction between these two components following the MEDIACTRL approach described in this chapter. Such adherence to the separation of concerns principle represents a first step towards a scalable system: in fact, we separated the signaling and conference management tasks (and the server-side of the additional functionality described in Section 2.4, too) from the handling of RTP streams. This last task is indeed very CPU intensive, as it encompasses media transcoding and mixing on a per user basis.

It is worth mentioning that a further step in the direction of a scalable system might be accomplished by implementing the *Media Resource Broker* role the MEDIACTRL working group is still working on [8]. Such entity manages the availability of Media Servers and the media resource demands of Application Servers when a M:N topology is envisaged.

Chapter 4

DCON: a scalable distributed conferencing framework

4.1 Introduction

The distributed conferencing architecture we defined has been conceived to be highly reliable and scalable. It has been called *DCON* [24], standing for *Distributed Conferencing*, but at the same time explicitly recalling the already mentioned XCON model.

DCON is based on the idea that a distributed conference can be setup by appropriately orchestrating the operation of a set of XCON focus elements, each in charge of managing a certain number of participants distributed across a geographical network. Interaction between each participant and the corresponding conference focus is based on the standard XCON framework, whereas inter-focus interaction has been completely defined and specified by us. In this chapter, we will describe a set of protocols we identified which are complementary to the call signaling protocols and are needed for supporting advanced conferencing applications, and we will provide the reader with some information about how we implemented the proposed distributed framework by extending the Meetecho platform described in Chapter 2.

4.2 Framework requirements

In order to build distributed conferencing on top of the already available centralized conferencing framework, we basically need to introduce additional functions:

- a coordination level among conference focus entities;
- a way to effectively distribute conference state information;
- some means to get centralized protocols to work in a distributed environment;
- a mechanism to distribute the media mixing process.

A more in-depth overview on these and other requirements is provided in Section 4.4.1.

The coordination level is needed in order to manage a distributed conference along its entire life-cycle. For instance, once a user decides to create a new conference, the corresponding conference focus has to distribute conference information to all other foci, so to enable other potential participants to retrieve the needed data and possibly subscribe to the event. We assume that all the operations needed inside a single conference "realm" are managed via the XCON protocols and interfaces, as implemented in *Meetecho*. Hence, each single realm keeps on being based on a star-topology graph for all what concerns the call signaling part. The various available stars are then connected through an upper-layer mesh-based topology providing inter-focus communication.

As to the second point mentioned above, it looks clear that a way to propagate information about conferences is needed when switching the view from a centralized to a distributed perspective. Indeed, whenever a new conference is created (or an active conference changes its state) such an event has to be communicated to all interested (or active) participants. Given the intrinsic nature of the distributed framework, the actual flow of information will always foresee the interaction among conference focus entities for both conference information exchanging and state changes notifications.

Conference state propagation can take place in a number of alternative ways. For instance, each focus might flood the received information across the inter-focus communication mesh, thus guaranteeing that potential participants belonging to heterogeneous islands can be reached. In such case, focus entities are *stateful*, i.e., each of them stores information about current sessions and forwards such information to all peering entities in order to get them up-to-date with respect to available conference sessions.

On the other hand, a distributed repository might be employed for the sake of storing conference information: focus entities would access such repository, both to publish (either upon creation of a new conference, or to notify a change in the state of an active conference) and to retrieve information about active conferences (e.g., when a new participant wants to access the list of ongoing/scheduled conference sessions he might be interested to join). In this last case, focus entities are *stateless*. Additional mechanisms for inter-focus interaction can be envisaged, the most interesting one being a pure peer-to-peer (P2P) approach.

4.3 Design

4.3.1 DCON framework

DCON has been conceived as a large scale evolution of our *Meetecho* framework. We deploy our architecture on top of a two-layer network topology (see Fig. 4.1). The top layer is represented by an overlay network in which each node plays the role of the focus element of an XCON “island”. The lower layer, in turn, is characterized by a star topology (in which the central hub is represented by the focus element) and is fully compliant with the XCON specification. In the DCON scenario, communication among different islands becomes of paramount importance. To the purpose, we chose to adopt the so-called S2S (*Server to Server*) module of the XMPP proto-

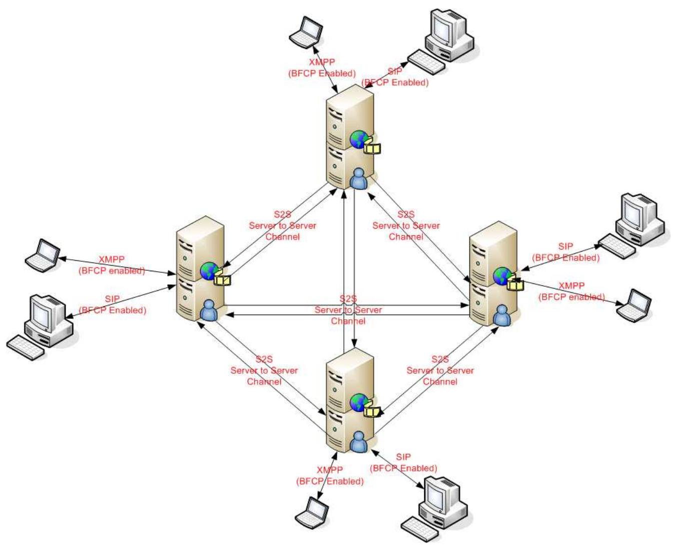


Figure 4.1: DCON design: the stateful approach

col. XMPP has been standardized by the IETF as the candidate protocol to support instant messaging, e-presence and generic request-response services, and it looked to us as the ideal communication means among DCON focus entities.

In the following section we will present a prototype solution based on the stateful approach.

4.4 Implementation

In this section we provide some useful information about the current implementation of the DCON framework. We will herein focus on a prototype which implements the DCON basic functionality by assuming a stateful scenario. Fig. 4.2 depicts the main implementation choices of the DCON architecture.

The right-hand side of the picture presents the logical view of the server, integrating our Asterisk-based *Meetecho* implementation of the XCON focus (upper box) with a brand new module specifically conceived for the SPreAding of Conference Events (which we called *SPACE*). SPACE has been realized as a plug-in for *Openfire* (lower box), a popular open source instant

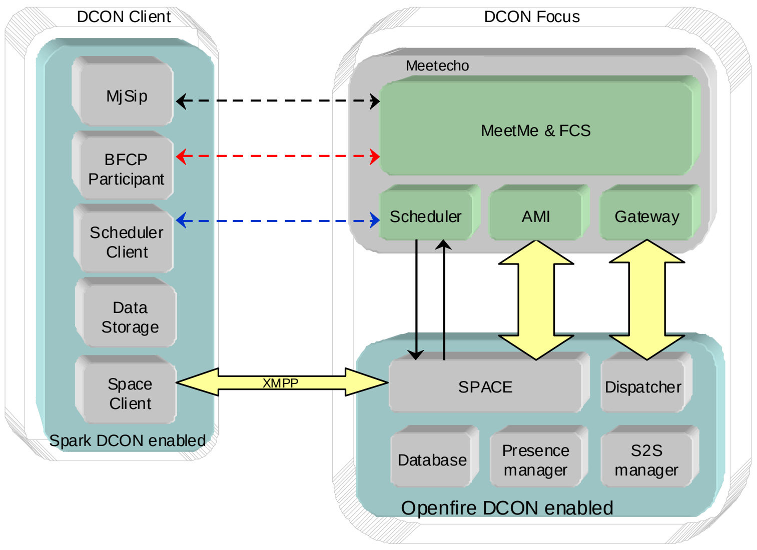


Figure 4.2: Implementation of the DCON architecture

messaging server already introduced in Chapter 2. We chose Openfire, since it comes with a native implementation of the above mentioned S2S channels. SPACE actually represents a key component of the architecture, since it enables inter-focus communication through the exchanging of conference information and forwards any natively centralized protocol message to all involved XCON clouds. In order to perform these two functions, the needed interaction between Openfire and Asterisk happens through both the already available Asterisk Manager Interface (AMI) and an ad-hoc protocol (called XDSP, *XCON-DCON Synchronization Protocol* [26]) that we developed from scratch. Inside DCON, communication between the legacy *Meetecho* modules and the newly created spreading components takes place on the basis of an asynchronous paradigm in which a number of events are generated by *Meetecho* modules whenever something relevant happens in the XCON island they are currently supervising. Furthermore, whenever a centralized protocol message is to be received by a different XCON cloud, it is forwarded from the *Gateway* module to the related DCON focus entity.

The left-hand side presents a logical view of the *DCON client*. It has been designed as an integrated entity realized extending the Meetecho client described in Chapter 2, and capable to interact with the framework by means of SIP (*MjSIP in the figure*) as well as BFCP (*Participant*), CCMP (*Scheduling Client*), and instant messaging (*SpaceClient*).

4.4.1 Inter-focus interaction

The SPACE module has been conceived to enable DCON functionality both on the server and on the client side. The following paragraphs will explain the respective roles in further detail.

Server side

On the server side, according to [25] we had to introduce support for the following new features:

1. Core functionality of the DCON focus:

   We concentrated our efforts on the integration between the Asterisk-based Meetecho server and the newly introduced DCON Openfire component. In order to offer a richer experience to users, we also implemented a direct communication channel with the Meetecho XCON scheduler for session management. Both the above mentioned features rely on the existence of a conference database, which we also implemented from scratch. The database contains relevant information regarding both active and registered DCON conferences;

2. Focus discovery:

   We chose to implement such new feature based on a client-driven IM presence service. Whenever the first client becomes active in a DCON cloud, the related focus entity opens an S2S channel towards all other focus entities it is aware of, thus triggering the spreading phase;

3. Information sharing:

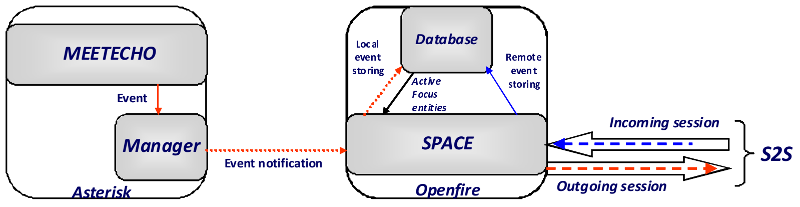


Figure 4.3: The spreading of conference information

Once the S2S channel has been setup as previously explained, the focus entities at both edges of the channel start sending each other their stored data. Data flowing along the S2S channel are generated by triggers coming from the associated Meetecho module. As depicted in Figure 4.3, whenever something relevant happens in an XCON cloud, Meetecho raises an event through the Asterisk manager interface. Such an event is then intercepted by the SPACE plug-in which – besides alerting all the associated clients – takes care of forwarding it across all established outgoing sessions.

4. Distributed conference management:

   As already stated, in order to allow users' access to remotely created conferences, we had to provide a mechanisms enabling transparent management of both locally- and remotely-created conference instances. We carried out this task by assuming that all conferences registered at the same focus are characterized by a common prefix. In such a way, the *Gateway* component of the focus is able to determine whether a request has to be processed locally or it has to be forwarded to the appropriate peer entity through the S2S channel.

5. Centralized protocols routing and dispatching:

   Having in mind the above mentioned assumption about conference prefixes, every time a centralized protocol message has to be dispatched, it is encapsulated in the payload of an ad-hoc XMPP packet and sent to the appropriate focus entity over S2S channels. Natively centralized protocol messages include, but are not limited to, any protocol defined and specified in the XCON framework. We will show in Appendix A how we coped with the dispatching of the BFCP protocol.

6. Distributed mixing:

   As soon as two or more centralized conferencing islands get connected in order to provide for a distributed conferencing scenario, we also cope with the issue of mixing media flows generated by the conference participants. In order to optimize the performance of the overall architecture, each focus is in charge of mixing the media streams of the island it supervises, and subsequently sending it over an RTP trunk to all the other foci involved in the scenario.

Client side

On the client side, we developed from scratch an integrated component that offers the following functionality: (i) retrieval and visualization of conference information; (ii) creation of a new DCON conference; (iii) joining (and unjoining) to an existing DCON conference; (iv) support to moderation.

In the following of this section we will briefly touch on all of the above mentioned points.

Regarding the visualization of conference information, we already showed how we provided the Spark client with a dedicated panel showing the details about both active and scheduled conferences (see Fig. 2.4 in Chapter 2).

As described in the previous section, upon activation of the first connection in an XCON cloud, the focus opens the S2S channels towards the peering entities and immediately thereafter sends the retrieved conference information to the connecting client. Moreover, if an active connection between the client and the focus is already in place, it is nonetheless possible for the client to asynchronously contact its reference focus in order to have up to date information about available conferences.

As to the joining of a conference hosted by a remote focus, we show in Fig. 4.4 the relevant information flow. As already said, in this case the inter-

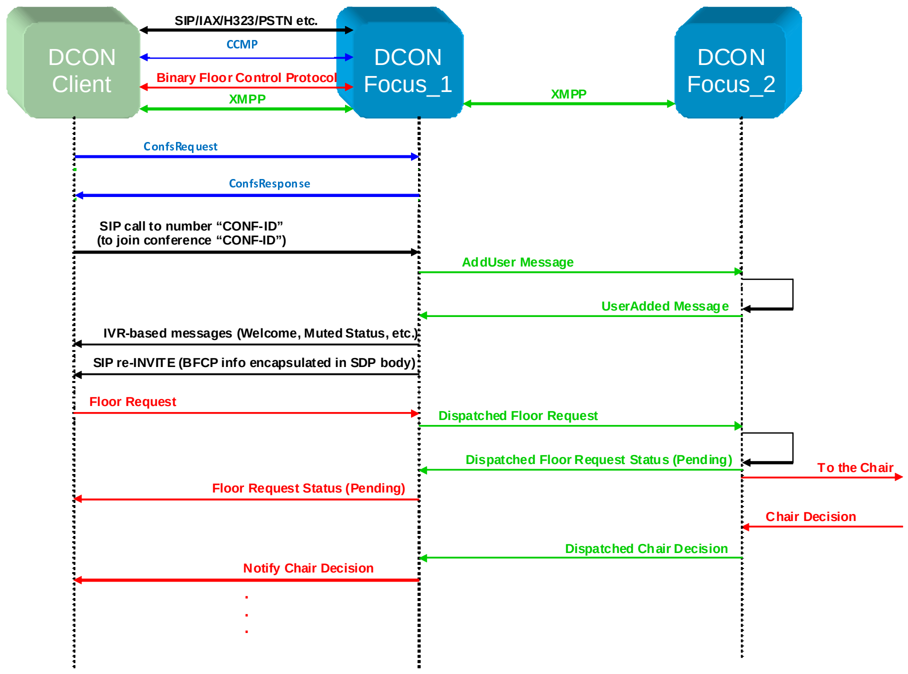

DCON Client
DCON Focus_1
DCON Focus_2
SIP/IAX/H323/PSTN etc.
CCMP
Binary Floor Control Protocol
XMPP
XMPP
ConfsRequest
ConfsResponse
SIP call to number "CONF-ID" (to join conference "CONF-ID")
AddUser Message
UserAdded Message
IVR-based messages (Welcome, Muted Status, etc.)
SIP re-INVITE (BFCP info encapsulated in SDP body)
Floor Request
Dispatched Floor Request
Dispatched Floor Request Status (Pending)
To the Chair
Floor Request Status (Pending)
Chair Decision
Dispatched Chair Decision
Notify Chair Decision

Figure 4.4: Join of a remote DCON conference

action between the client and its focus does not change. The only difference consists in the fact that behind the scenes a number of messages are exchanged between the involved foci (i.e., the client's reference focus on one side and the focus supervising the remote cloud where the conference was originally created on the other side) in order to effectively add the user to the conference and correctly generate the re-invite message.

Appendix A

BFCP dispatching in DCON

A DCON compliant architecture must provide users with the capability to transparently join a conference which is hosted by a focus entity belonging to a foreign centralized island. Furthermore, it must forward any centralized protocol message to its related peer in the distributed overlay whenever the message is directed to a receiver who does not belong to the local centralized system. Our implementation of the DCON system fully complies with the mentioned requirements. In fact, in order to join a remote conference the participant only needs to interact with its corresponding focus in exactly the same way as he did in the centralized case. Upon reception of the participant's request, the local focus forwards join information to the focus entity belonging to the island in which the conference in question was created. Due to the large number of peer entities each focus could be in touch with, in a distributed scenario a mechanism to prevent collisions in the assignment of centralized protocols identifiers is needed. We achieved this goal by means of the label swapping process explained with the following example.

To make the whole thing clearer, we deal with the case where a user (Client (A)) belonging to XCON cloud (A) wants to remotely participate in a distributed conference hosted by XCON cloud (B). Fig. 4.5 shows how this kind of join requests is handled by the local focus in order to both dispatch them and create a sort of *virtual labels path* needed to correctly manage future messages addressed to the same user in the same conference.

1. Once the client has locally joined the distributed conference by placing a SIP call to the focus it belongs to (XCON (A)), the focus chooses a new label for Client (A) which will be used to appropriately dispatch all messages related to it;

2. XCON (A) at this point forwards the join request to its related DCON focus entity (DCON (A)); in this example this is done by sending, through the *XCON-DCON Synchronization Protocol (XDSP)*, a message called *AddUser*, containing the newly assigned client's label A;

3. DCON (A) receives the join request; since it regards a new client, the DCON focus entity chooses a new label (e.g., XYZ) and associates it with the just received label A; depending on the distributed conference the client wants to join, it associates the label (XYZ) with the DCON focus entity managing the XCON focus physically hosting the conference (DCON (B)) and forwards the join request to it;

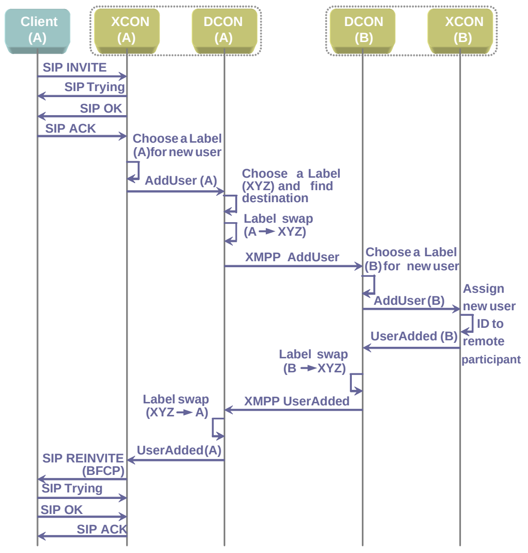


Figure 4.5: Joining a remote conference

4. DCON (B) receives the forwarded message through a *server-to-server* (s2s) channel based on the XMPP protocol; since it regards a new client, DCON (B) chooses a new label (e.g., B) and associates it with the just received label XYZ; since the conference the remote Client (A) wants to join is physically hosted by XCON (B), the join request is forwarded there using the XDSP protocol, with an *AddUser* message containing the newly assigned label B which identifies the remote client;

5. XCON (B) receives the request, and thus associates the received label B with the remote Client (A); all the operations needed to add the new user to the conference are performed, and the new information is sent back to the client through the same path. All the involved labels (B, XYZ, A) will be properly swapped to route all XCON protocols messages between the two entities.

Once XCON (A) receives the confirmation that the user has been successfully added to the remote conference, together with the related information, the Client (A) is updated through a *SIP re-INVITE* containing the BFCP information needed to communicate with the Floor Control Server.

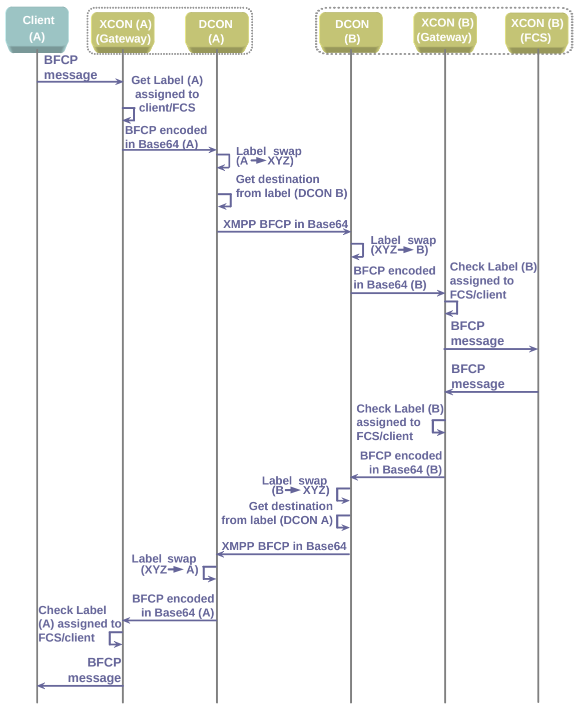

Client (A)
XCON (A) (Gateway)
DCON (A)
DCON (B)
XCON (B) (Gateway)
XCON (B) (FCS)
BFCP message
Get Label (A) assigned to client/FCS
BFCP encoded in Base64 (A)
Label swap (A → XYZ)
Get destination from label (DCON B)
XMPP BFCP in Base64
Label swap (XYZ → B)
BFCP encoded in Base64 (B)
Check Label (B) assigned to FCS/client
BFCP message
BFCP message
Check Label (B) assigned to FCS/client
BFCP encoded in Base64 (B)
Label swap (B → XYZ)
Get destination from label (DCON A)
XMPP BFCP in Base64
Label swap (XYZ → A)
BFCP encoded in Base64 (A)
Check Label (A) assigned to FCS/client
BFCP message

Figure 4.6: BFCP dispatching

From this moment on and for the entire life cycle of its participation in the remote conference the local focus takes care of the user acting as a gateway for all what concerns natively centralized protocols.

In order to make the reader as aware as possible of the mechanisms involved in the experimental campaign presented in the following sections, Figure 4.6 illustrates how BFCP is dispatched across the distributed environment.

Continuing from the previous example, since XCON (B) is physically hosting the conference, floor control will be entirely managed by its Floor Control Server. All BFCP messages sent by the clients to the Floor Control Server are intercepted by the local focus (specifically by its *Gateway* component), and then forwarded to the Floor Control Server of XCON (B). We have already seen how labels are assigned and swapped: the same labels will be used for dispatching.

The flow of a typical message exchange can be seen as follows:

1. Client (A) sends a BFCP message to the Floor Control Server; the message is intercepted by XCON (A)'s gateway; the label assigned to client (A) is retrieved, and used to forward the BFCP message to the Dispatcher in DCON (A); of course, since BFCP messages are binary, a proper Base64 encoding is performed to encapsulate the message in our text-based XDSP protocol data unit;

2. Once DCON (A) receives the encapsulated BFCP message, the labels are once again swapped (in this case, A $\rightarrow$ XYZ) and the message is routed to the right destination (DCON (B));

3. DCON (B) will receive the message and swap labels again (XYZ $\rightarrow$ B); at this point, the encapsulated message will be forwarded to the underlying XCON (B) Gateway to be further processed there;

4. The Gateway in XCON (B) will receive the encapsulated (and Base64-encoded) BFCP message; after decoding it, the Gateway will analyze the label marked in the message (B, in this case), and will understand it is a message sent by a remote user (Client (A)) to the local Floor Control Server. It will forward the (now 'natively' binary) message there, where it will be processed;

5. In case FCS (B) needs to send a message to Client (A), exactly the same operations will be performed, and the same path will be followed through the needed label swaps among the involved peers. FCS (A), while not actually managing the floors related to the remote conference in which Client (A) is participating, will however be notified upon each floor status change, so to appropriately update the local media mixes when needed (e.g., to mute Client (A), thus excluding her/him from XCON (A)'s local mix if FCS (B) has decided so).

Chapter 5

From theory to practice: a scalability analysis

5.1 Preliminary considerations

The objective of this chapter is to demonstrate how the migration from the Meetecho centralized platform described in Chapter 2 to the DCON distributed environment (see Chapter 4 can improve the scalability of the overall system. The performance tests we carried out focused on two aspects:

- the number of users the system is able to manage;
- the CPU load and hardware resources consumption given a certain amount of users in the system.

In both cases we exploited the SIPp[1] tool, an open source performance testing tool for the SIP protocol that generates traffic according to a fully customizable XML scenario file. While it includes some basic template files for the most common call scenarios, we nonetheless had to create our own in order to manage the re-INVITE message sent back from our Meetecho server, together with the subsequent replies that constitute the second three way handshake characterizing the joining of a conference. Another useful feature of SIPp is the possibility to send media traffic (both audio and video)

[1]`http://sipp.sourceforge.net`

through RTP/pcap replay, thus not limiting the tests only to call signaling related matters. Using SIPp, we were hence capable to stress both the centralized and the distributed platform. Though, in order to involve all the components of the architecture in our analysis (including the Media Server and the Floor Control Server), we also had to implement a “stresser” for the BFCP protocol. Such a tool is capable to send *FloorRequest* and *FloorRelease* BFCP messages to the FCS, thus enabling or disabling the mixing operation for each call instance. On the other hand, when no BFCP functionality is involved in the performance tests, the media mixer would not be involved in the scenario at all, as the media streams coming from the conferencing clients would just be discarded.

In the following, we first provide a brief description of the *BFCP stresser* we implemented; then, we show the experimental results obtained when testing the Application Sever alone (i.e., with no BFCP involvement) as well as when involving the Media Server and the Floor Control Server.

The experimental campaign we conducted is based on the realization of a great number of tests for a single scenario. This is done in order to avoid basing our analysis on a potentially biased set of data. For each scenario, we present a single result which is representative of the average case for that scenario.

5.1.1 The BFCP stresser

As already mentioned, we wanted to involve in our measurement campaign both the media mixer and the Floor Control Server, so the need arose for a tool capable to send BFCP messages on behalf of the participants of a conference. We remind that in order to build a BFCP message, we need the transport address of the FCS, together with the related identifiers. Such information is encapsulated in the SDP body of the re-INVITE message the focus sends back when a user joins a conference. In spite of this, we decided to statically provide our tool with these data by means of a configuration file, since we did not want to force the *BFCP stresser* to be co-located with the

SIPp stresser (which is in charge of signaling tasks). Furthermore, in order to build a BFCP message, the *BFCP stresser* has to be aware of user-specific parameters (i.e., UserID and ConferenceID). In our implementation, these parameters are extracted from an asynchronous event raised by the Asterisk server whenever a new user joins a conference. So, we let our tool connect to Asterisk through the *Asterisk Manager Interface*, making it capable to catch any suitable event and act consequently. We remark that the software developed also maintains information about each user's state, saving the related floor status (granted/not granted). This is useful since we envisaged the possibility of setting two parameters, p_{req} and p_{rel}, related to the probability per second with which the tool generates, respectively, *FloorRequest* and *FloorRelease* messages on behalf of each user, according to her/his status. For example, if we set $p_{req} = 0.15$ and $p_{rel} = 0.05$, every second the stresser checks each user's status and:

- if she/he has not been granted the floor, the tool performs a *FloorRequest* with a 15% probability;
- if she/he has already been granted the floor, a *FloorRelease* is performed with a 5% probability. The desired values of these probabilities are to be specified through the already mentioned configuration file.

5.2 Stressing the Application Server

5.2.1 Centralized scenario

Fig. 5.1 illustrates the testbed configuration in the case of the centralized conferencing scenario. The *SIPp console* is used to remotely control and coordinate a single test, by issuing commands to the call originating server (*SIPp stresser* in the picture). Such server in turn creates and sends call requests to the Confiance server in accordance with the SIPp console indications, thus emulating a real-world calls profile. During a test lifetime, the logging database is continuously updated with information about the Meetecho server status. We then exploit the gathered information in order to

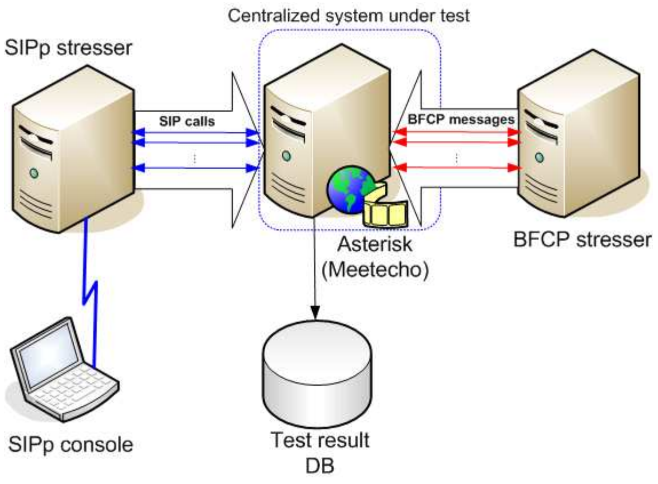


Figure 5.1: Testbed configuration: centralized scenario

perform off-line analysis of the collected data and evaluate the results. We remark that all the server machines used for the testbed are hosting a Linux Debian 4.0 “Etch” operating system equipped with a $2.6.18-5-686$ kernel. They have a $3.2GHz$ Intel XEON CPU and $2GB$ RAM. Bandwidth was not a limiting factor for the performance evaluation, since we used a dedicated gigabit ethernet network for the campaign.

In this phase, when testing the AS to evaluate the number of supported users, we did not make use of the *pcap replay* function of SIPp. This was done in order to guarantee that each client only generated signaling traffic without sending any RTP packet to the focus. We were thus able to point out an intrinsic limitation of the Asterisk server, that does not allow to have more than 1024 Zaptel [2] pseudo-channels open simultaneously. For the sake of completeness, we remark that in order to reach the above mentioned limit we had to modify the number of open file descriptors the operating system allows to have, setting it to its maximum value of 65535. These Zaptel pseudo-channels are needed by the MeetMe application module for creating a shared “bus” for the management and transport of the mixed audio related to the conference. Furthermore, they are also used for realizing

[2]Jim Dixon’s open computer telephony hardware driver API used by Asterisk.

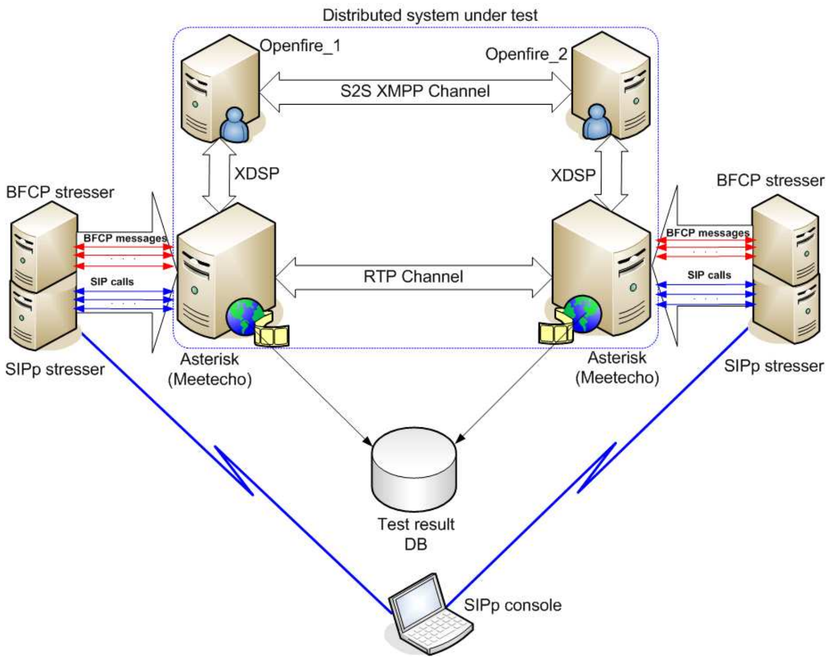


Figure 5.2: Testbed configuration: distributed scenario with just two XCON islands

so-called “listeners” exploited by users in order to get attached to the above mentioned channel. This kind of listeners are the same MeetMe exploits to make audio recordings of running conferences, when needed. Because of this limitation, and supposing that there is only one active conference with no active recording running in the background, it is not possible to have more than 1022 users. In fact, as soon as a conference is created and becomes active, two channels are used by MeetMe itself, while other channels are used when new users join the conference.

5.2.2 Distributed scenario

Fig. 5.2 illustrates the testbed configuration in the case of the distributed conferencing scenario with just two interconnected conferencing islands.

With respect to the centralized case, we now find the *Openfire* component, which is specifically dedicated to information spreading and protocol dispatching. We also notice that the *SIPp console* is now used to manage both of the available *SIPp_stresser* entities. In the following of this subsection we will present results obtained in the distributed scenario, for an increasing number of interconnected islands. Namely, we will analyze in detail the results obtained, respectively, with two, three and four conferencing clouds. We are not providing any illustration of the three- and four-elements scenarios, which do not present any new feature when compared to the above depicted two-elements case.

Coming to the tests, we have thoroughly explained in Chapter 4 how all the users behind a remote focus are seen from the main focus. While their signaling and control is managed through protocols dispatching and inter-focus synchronization, their media (both audio and video) are locally mixed on the remote focus, and then sent to the main focus as if they came from a single user. The same thing obviously happens for the media flowing from the main focus towards the remote foci as well. The main focus locally mixes the media coming from its local users and from all the remote foci involved in the conference, to which it forwards them. So, since only one RTP trunk per medium is needed between each remote focus and the main focus, we expected the maximum number of participants to linearly grow with the number of "DCON islands". This statement was promptly confirmed by our tests in all cases (two-, three- and four-island topologies). It is important to notice that each remote focus attached to the main focus means a new listener on the shared bus channel, and so one less available channel for local users. This last statement is true for both the main and the remote focus. Fig. 5.3 summarizes the above considerations, by showing the linear increase in the number of supported users in the four scenarios considered.

5.3 Involving the Media Server

5.3.1 Centralized scenario

When involving the media mixer in our scalability measurements, we focused on the resource consumption aspect of the scalability. We found that the bottleneck is definitely represented by the CPU utilization level, rather

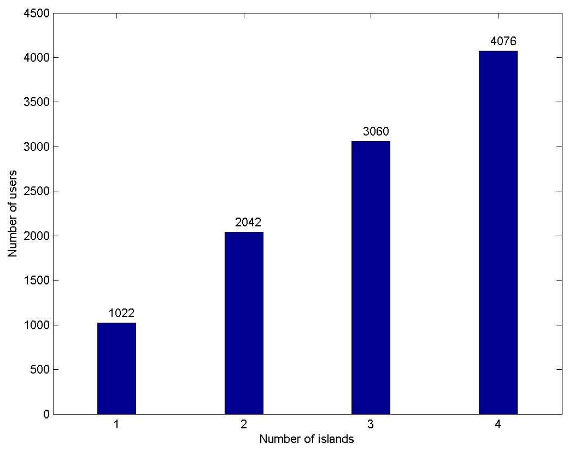


Figure 5.3: Number of users supported in the four analyzed scenarios

than other relevant hardware parameters like, for example, available RAM memory. In the following, we will hence focus just on CPU as the key performance indicator. In this phase the *BFCP stresser* component of the testbed comes into play, which hosts the tool we developed in order to generate BFCP messages and consequently allow/deny the mixing of the media flows related to each call. We made also use of the *pcap replay* functionality of SIPp, so that each user sent both SIP and RTP traffic. Fig. 5.4 shows the CPU utilization level in the presence of only one active conference and considering all the participants to have been granted the audio floor. This behavior was obtained by appropriately instructing the *BFCP stresser* to send a *BFCP FloorRequest* on behalf of each user who joined the conference, and to never release the obtained floor. It can be seen that when the number of participants approaches 180 the CPU load of the focus managing the conference is close to 100%.

The result of 180 users as the peak value for the centralized scenario in the presence of BFCP functionality will be used as a benchmarking parameter in the following subsection, which discusses the achievable performance improvement in the distributed case.

It is worth noting that the results presented above, and in the following as well, do not strictly depend on the assumption that all users participate

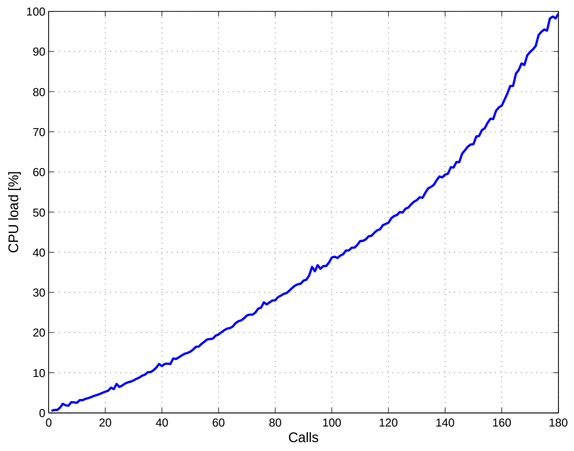


Figure 5.4: CPU utilization in the centralized scenario with moderation

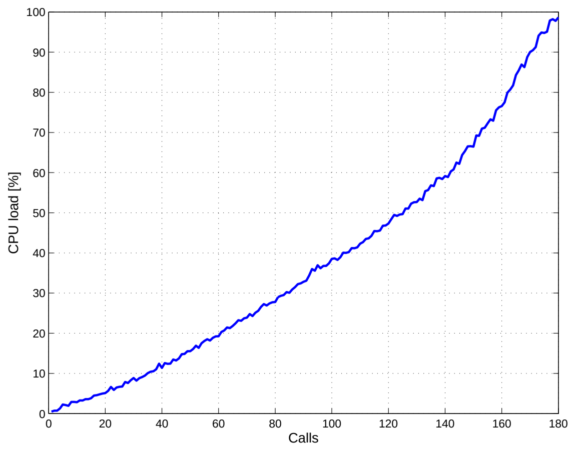


Figure 5.5: CPU utilization in the centralized scenario with moderation, in the presence of 10 active conferences

in the same conference. To confirm this, we considered a more realistic scenario characterized by 10 active conferences with 18 participants each. This situation led to identical performance figures as shown in Fig. 5.5.

5.3.2 Distributed scenario

As already said, we wanted to study how the CPU performance improved when moving from a centralized to a distributed environment. With this objective in mind, we performed several testing experiments, assuming, as in the centralized case, the presence of just one active conference with 180

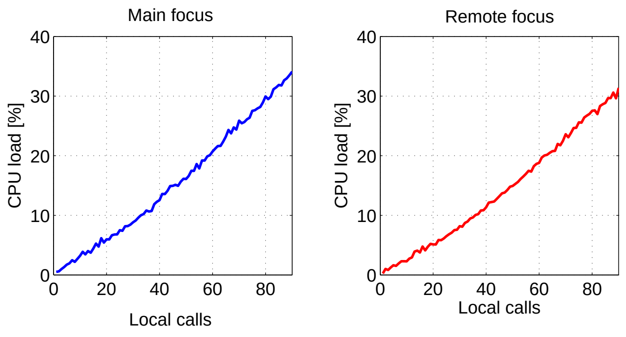


Figure 5.6: CPU utilization in the distributed scenario envisaging the presence of two XCON islands

participants. Specifically:

1. we first considered a two-islands topology with 90 local users and 90 remote users;
2. we then moved to a three-islands scenario with:
   (a) 60 local users and 120 remote users, equally split between the two available remote foci;
   (b) 90 local and 90 remote users, equally split between the two available remote foci.

In the first scenario, the CPU load of the main focus was about 34%, while the remote focus was about 30%, as shown in Fig. 5.6.

In case 2a, instead, the main focus CPU level was about 21% and the remote foci were both loaded around 19%. This is shown in Fig. 5.7.

Finally, in case 2b, the main focus took up 32% of the CPU, while the remote foci were almost unloaded (CPU utilization level at around 13%). This is witnessed by Fig. 5.8.

5.3.3 Comparative analysis

In this sub-section, we analyze the results presented above, showing (see Fig. 5.9) how the migration from a centralized to a distributed paradigm

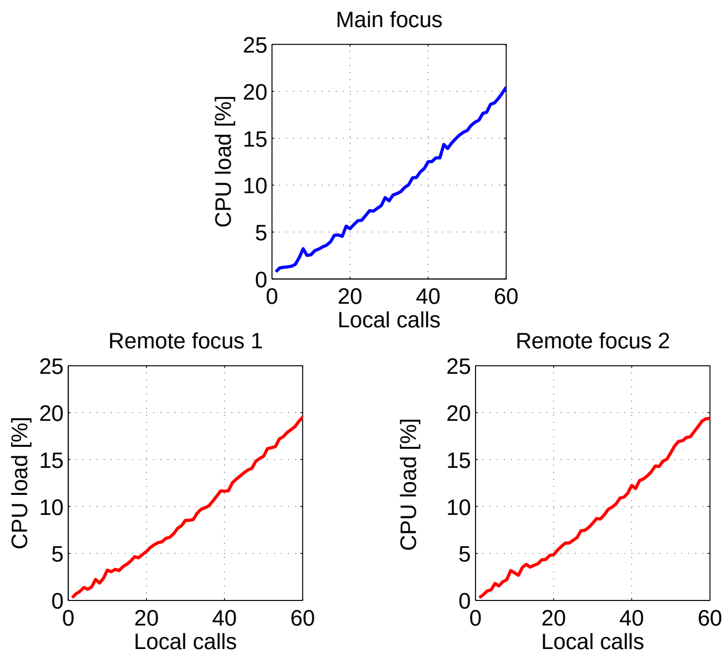


Figure 5.7: CPU utilization in the distributed scenario envisaging the presence of three XCON islands: case 2a

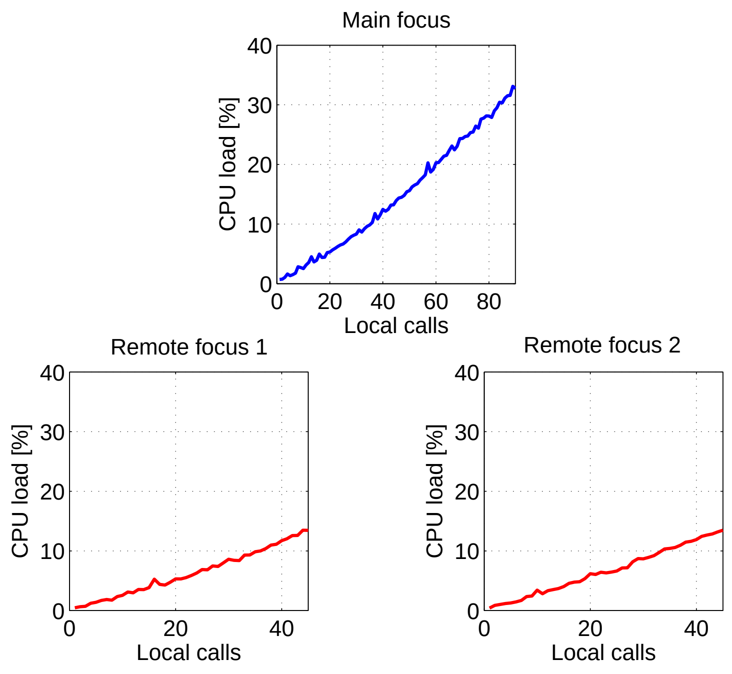


Figure 5.8: CPU utilization in the distributed scenario envisaging the presence of three XCON islands: case 2b

Number of islands	Number of local users	Number of remote users	Main focus CPU load	Remote focus 1 CPU load	Remote focus 2 CPU load
1	180	-	˜ 100%	-	-
2	90	90	34%	30%	-
3	60	120 (60/60)	21%	20%	20%
3	90	90 (45/45)	32%	13%	13%

Figure 5.9: A comparative table summarizing performance figures in the presence of BFCP

considerably improves the performance in terms of scalability. We have already shown the linear growth in the number of users related to the number of involved DCON islands. What we want to remark here is how our tests showed the huge improvement in terms of CPU performance, since in the centralized scenario examined, the CPU load was near 100% while it reduced to about 34% in the two-islands case and to 21% in the three-islands scenario. Furthermore, we notice that the sum of the CPU loads of all the foci involved in the various scenarios was also lower than the load of the centralized platform. Finally, as shown by experiments presented in case 1 and case 2b, we note how the distribution of the remote users behind different remote foci causes a small decrease in the performance of the main focus. This is reasonable since we know that the main focus has in such cases to spread conference events to all the active foci on the server-to-server channels.

5.4 Involving the Floor Control Server

In this subsection, we present the results obtained when the system under stress also included the Floor Control Server. To do this, we had to modify the previous test configuration scenario where a *FloorRequest* message was sent every time a new user joined a conference. During this campaign, instead, we configured the *BFCP stresser* with two parameters, p_{req} and p_{rel}, representing, respectively, the desired probability per second with which the stresser generates *FloorRequest* and *FloorRelease* messages on behalf of each participant. In the following, hence, we analyze the behavior of the system by assuming different values for those probabilities. We also point out the

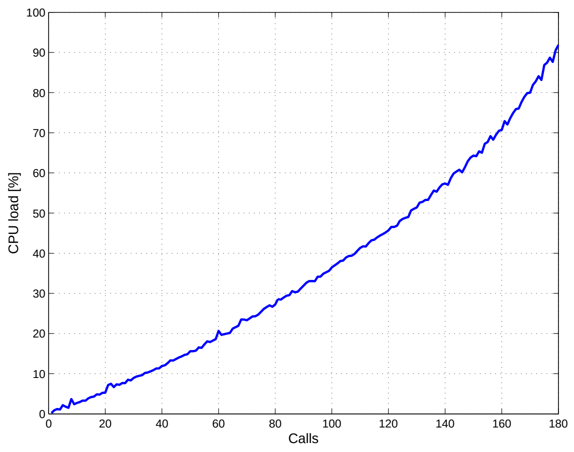


Figure 5.10: CPU utilization in a centralized scenario characterized by $p_{req} = 0.15$ and $p_{rel} = 0.05$

additional load due to moderation operations.

5.4.1 Centralized scenario

Starting from the centralized environment, we show the CPU utilization level obtained when fixing two different values for the probability parameters previously introduced. Specifically, we first assumed $p_{req} = 0.15$ and then $p_{req} = 0.30$, while in both cases the probability (per second) with which each participant performed a *FloorRelease* was set to 0.05. Such values just reflect the actual behavior inferred from empirical experiments previously arranged, and consequently represented a reasonable choice for our tests.

As to the first scenario, Fig. 5.10 shows that the CPU load of the focus when managing 180 users was about 92%. This reduction in the load when compared with the case of all participants owning the floor, can be explained by observing that in this case the focus did not have to mix all the incoming media flows, but just a subset of them.

Fig. 5.11, instead, witnesses how the same overall number of users in the second scenario (characterized by a higher floor request probability) almost overloads the CPU (whose utilization level reaches 99%).

The figures just presented bring us to some consideration about the con-

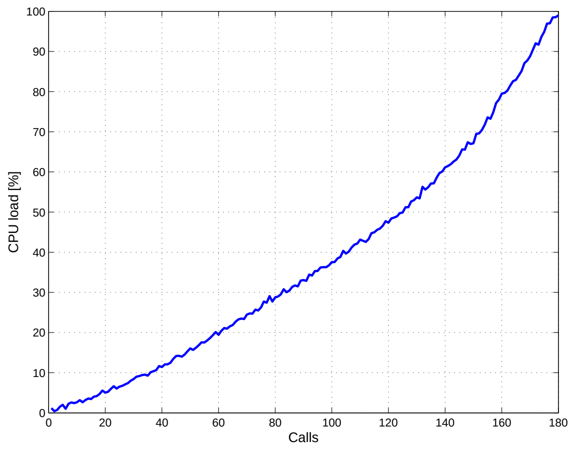


Figure 5.11: CPU utilization in a centralized scenario characterized by $p_{req} = 0.30$ and $p_{rel} = 0.05$

sumption of resources specifically and exclusively due to the activity of the Floor Control Server, regardless of any other task carried out by the focus. In fact, the scenarios above can be described through the very simple Markov chain depicted in Fig. 5.12, representing the state transitions of a single participant. From the analysis of that Markov chain, it follows that each user owns the audio floor with a probability of $P_{floor} = p_{req}/(p_{req} + p_{rel})$. Consequently, in the first scenario considered there are on average 135 audio flows to be mixed (which are related to participants owning the floor) in spite of the overall 180 users. Then, since subsection 5.2.1 showed us that 135 users owning the floor take up about 56% of the CPU and as long as 45 users without the floor require a further 8%[3], it is straightforward that the resource consumption specifically due to moderation operations is about 24%.

5.4.2 Distributed scenario

Coming to the distributed case, for the sake of brevity we just focus on the scenario characterized by $p_{req} = 0.15$ and $p_{rel} = 0.05$. Fig. 5.13 shows that in the case of two interconnected islands and supposing the users to be equally

[3]Figures related to scenarios where users send media traffic without owning the corresponding floor are not presented in this chapter. The interested reader might refer to [2], where a complete study of such cases is provided.

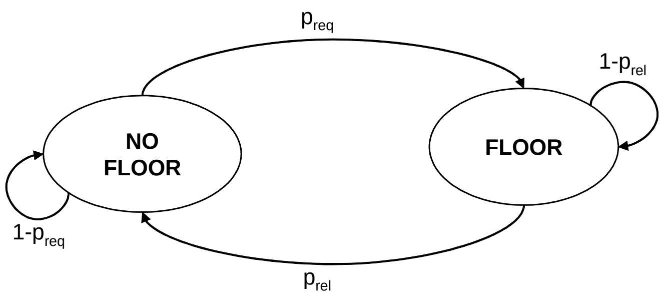


Figure 5.12: A Markov chain which describes moderated conferencing scenarios

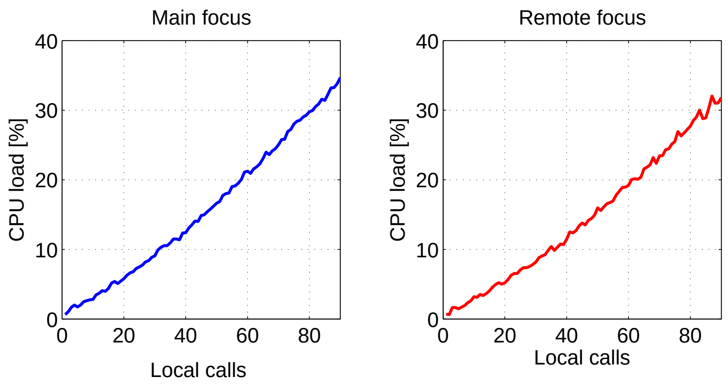


Figure 5.13: CPU utilization in a two-islands scenario characterized by $p_{req} = 0.15$ and $p_{rel} = 0.05$

split between them, the CPU level of the main focus was about 33% while the remote focus was loaded at around 32%.

Keeping on considering the participants equally split among the available domains, the three-islands case was characterized, instead, by 21% CPU consumption at the main focus and 20% at both remote foci (see Fig. 5.14).

In this subsection we do not deal with a comparative analysis between the centralized and the distributed environment, since the same considerations as in subsection 5.3.3 totally apply here.

Rather, we want to highlight how in the considered configuration of tests, working with average values becomes really crucial. In fact, when a scenario is characterized by a number of participants owning the floor that varies in time, the CPU occupancy given a certain amount of users varies too. This statement is confirmed by Fig. 5.15, showing the variation in time of the

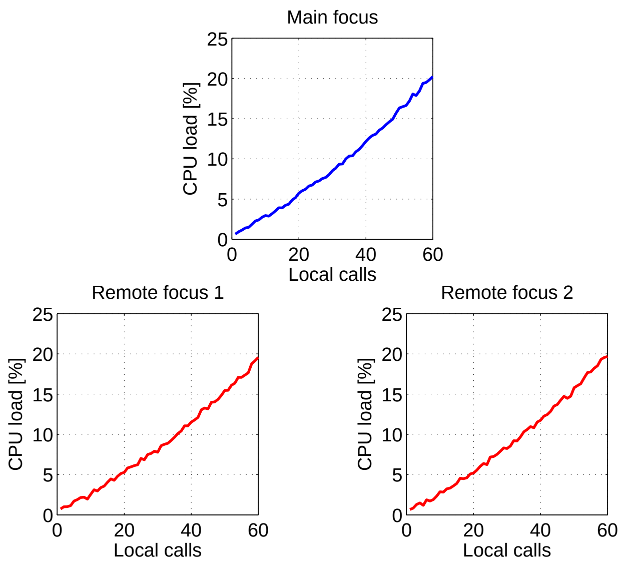


Figure 5.14: CPU utilization in a three-islands scenario characterized by $p_{req} = 0.15$ and $p_{rel} = 0.05$

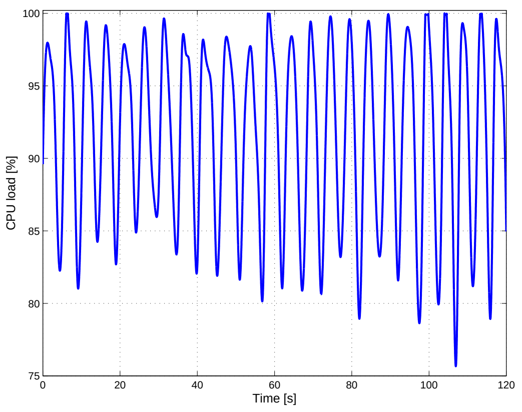


Figure 5.15: Time evolution of the CPU level in a centralized scenario characterized by $p_{req} = 0.15$ and $p_{rel} = 0.05$, when there are 180 users

CPU level given a fixed number of participants, in a single-island case with $p_{req} = 0.15$ and $p_{rel} = 0.05$. Such a variable CPU level depends on the actual number of audio flows that have to be mixed, since the less are the users able to talk, the less is the load of the focus and vice versa.

5.5 Considerations

The work carried out and presented in this chapter was mainly focused on a scalability assessment of the DCON framework in the presence of audio mixing operations and floor control. The results obtained and discussed in the previous sections show that our efforts in defining and implementing a distributed framework, with the corresponding issues about both distributed mixing and distribution of the floor control protocol, were well directed. Furthermore, the stressing operations conducted during this test campaign also proved very useful in finding any bug that may be hidden in the code and consequently affect the expected behavior of the platform. Finally, with respect to floor control, the performance figures we derived confirm the intuitive expectation that nothing comes for free. Moderation has to be paid for, since it does represent an expensive task, especially on the server side, where state information has to be kept and updated with reference to each and every user who is participating in a conference.

Chapter 6

Troubleshooting

6.1 Introduction

This chapter is devoted to troubleshooting issues we had to face when we decided to deploy our multimedia conferencing system in the actual Internet, thus bringing the results of our research activities from the lab to the real world. In fact, in order to let a complex and effective deployment of multimedia conferencing scenarios work properly, there is the need for a "co-operative" network, given the strong requirements they impose in terms of full access to the underlying network topology and resources. Unfortunately, this is often not the case for several different reasons. In the following, we will describe the issues we encountered and how we managed to identify and solve them by means of a network diagnosis architecture and a protocol tunneling solution, respectively.

6.2 Issues

6.2.1 Signaling plane

We explained how our architecture leverages the SIP protocol in the signaling plane. Such protocol, though it is the *de-facto* standard for instantiating multimedia sessions over the Internet, is very sensitive to the presence of middleboxes along the signaling path, like Network Address Translators (NATs).

In fact, when the SIP protocol has been specified by the IETF community, in the first place they neglected the issues NAT elements might cause, being confident that a full switch from IPv4 to IPv6 was forthcoming. They stated that a User Agent (UA) must send the media to the transport addresses indicated in the SDP body, even if they are received from another IP address and/or port. In scenarios where "natted" parties are involved, this would clearly cause RTP packets to be sent to a private, non-routable, IP address and to not be received form the other party. As the deployment of NAT boxes became larger, a lot of efforts have been done in order to address such issue. Within the IETF, the BEHAVE (Behavior Engineering for Hindrance Avoidance) Working Group deals with the behavior of NATs and has completed the definition of the STUN (Session Traversal Utilities for NAT) [14] protocol, and is still working the TURN (Traversal Using Relay NAT) [15] and ICE (Interactive Connectivity Establishment) [28] protocols, too. Such NAT-traversal techniques try to make a UA send the media flows to the correct transport address so that the remote party is able to receive them. They operates in different ways: STUN learns a node's *reflexive transport address*[1] and put it into the SDP body of SIP messages and into SIP headers, too; TURN makes use of a relay agent to/from which a natted node sends/receives the RTP streams; ICE combines the use of both STUN and TURN in order to cross each possible type of NAT. Unfortunately, even if there are lots of cases where the listed solutions work flawlessly, in some scenarios where very restrictive NATs are in place they are not able to do the job. In these cases a different solution has to be exploited, and server-side techniques are often the best choice. Meetecho, for the purpose, makes use of the *Asterisk SIP NAT solution*, which consists in ignoring the address information in the SIP and SDP headers from this peer, and replying to the sender's IP address and port. Furthermore, Asterisk ignores the IP and port in the SDP received from the peer and will wait for incoming RTP: after that, it already knows where to send its RTP.

[1]From RFC 5389: the *reflexive transport address* is the public IP address and port created by the NAT closest to the server (i.e., the most external NAT)

6.2.2 Media plane

The issues presented in the previous subsection, while related to the signaling plane, have strong repercussions on the media plane, as they result in the impossibility to establish a bi-directional communication channel between a caller and a callee. Moreover, problems in the media plane can also be due to devices included in the media path, be them RTP proxies, Application Level Gateways (ALGs) or NATs. Hence, problems in the media plane are hard to be diagnosed, as a number of different yet often interdependent causes might be the source. Section 6.3 will deal with fault diagnosis in networks based on the SIP protocol, also presenting a very common RTP fault as a representative scenario.

6.2.3 Proxy/Firewall traversal

It is not unusual that enterprises, work places, public entities like schools and universities, as well as other secured environments, aim at limiting or filtering out the traffic flowing across their networks. These limitations are usually enforced by deploying more or less restrictive elements like firewalls or proxies at the borders of the network topology, which limit the allowed traffic according to a specific set of policy rules. Such rules may include port- and/or domain-based filtering, filters on TCP or UDP traffic, payload inspection, limitations on the type and number of supported protocols, etc. It is not unlike to meet restrictive elements that only allow HTTP and/or HTTPS traffic to pass through, with the HTTP traffic itself that might also be subject to further, payload-based inspections. These scenarios are not new at all, and solutions to effectively deal with them have already been proposed and exploited with alternating success during the years. Specifically, the most common approach that is exploited to get around such limitations is HTTP tunneling. Such approach raises issues when exploited to carry protocols that are sensitive to transport-related information, like the ones involved in multimedia conferencing scenarios. In fact, as we will show in Section 6.4, we had to implement a custom Meetecho-aware tunneling solution in order to get around proxies and firewalls.

6.3 DYSWIS: A distributed approach to network diagnosis

In this section we present an architecture for network fault diagnosis currently under development at the IRT Lab of Columbia University in collaboration with the COMICS Lab of the University of Napoli Federico II. Such architecture, called DYSWIS (Do You See What I See?) leverages distributed resources in the network, called *DYSWIS nodes*, as multiple vantage points from which to obtain a global view of the state of the network itself. Each DYSWIS node is capable to detect fault occurrences and perform or request diagnostic tests, and has analytical capabilities to make inferences about the corresponding causes.

6.3.1 Architecture

From a very high-level perspective, a DYSWIS node tries to isolate the cause of a failure by asking questions to peer nodes and performing active tests. The architecture is depicted in Fig. 6.1. A modular approach is adopted, in order to allow support for new protocols in an easy fashion. Specifically, each time a new protocol has to be added, protocol-specific *Detect* and *Session* modules have to be implemented, together with a representation of the fault. Furthermore, new tests and probes have to be implemented, too, when required. Finally, the rules that drive the diagnosis process have to be written. In fact, each DYSWIS node relies on a rule engine that triggers the invocation of the probes on the basis of the type of fault and of the result of previous tests.

As probing functions need to be executed on remote nodes that have specific characteristics, a criterion to identify such nodes is needed, as well as a communication protocol. For example, we might be interested in selecting a peer that has a public IP address, rather than a node that belongs to a

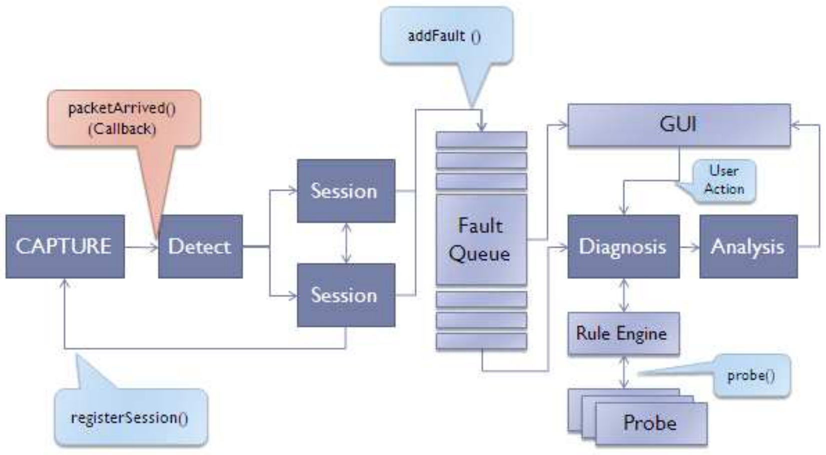


Figure 6.1: DYSWIS architecture

given subnet. At the time of writing, remote peers are discovered by means of a centralized repository where each node registers all its useful information as soon as it becomes available. However, an alternative approach, exploiting a *Distributed Hash Table* (DHT), has been implemented in order to better fulfill scalability requirements.

In order to communicate among each other, as well as to convey information about detected failures and request a probe to be run, the DYSWIS nodes exploit a request-response protocol. For further details about how this functionality is provided, refer to Section 6.3.4, which discusses implementation aspects.

Finally, when the probing phase is completed, the *Analysis* module produces the final response and presents it to the user.

6.3.2 Diagnosing SIP/RTP faults

As already introduced, our purpose was to exploit the DYSWIS framework to address the issues presented in Section 6.2.2. For this purpose, we had to extend the features provided by the framework, as it did not provide support for VoIP protocols. Hence, we added support for both SIP and RTP. The detection part is simply performed by "sniffing" packets on the SIP standard

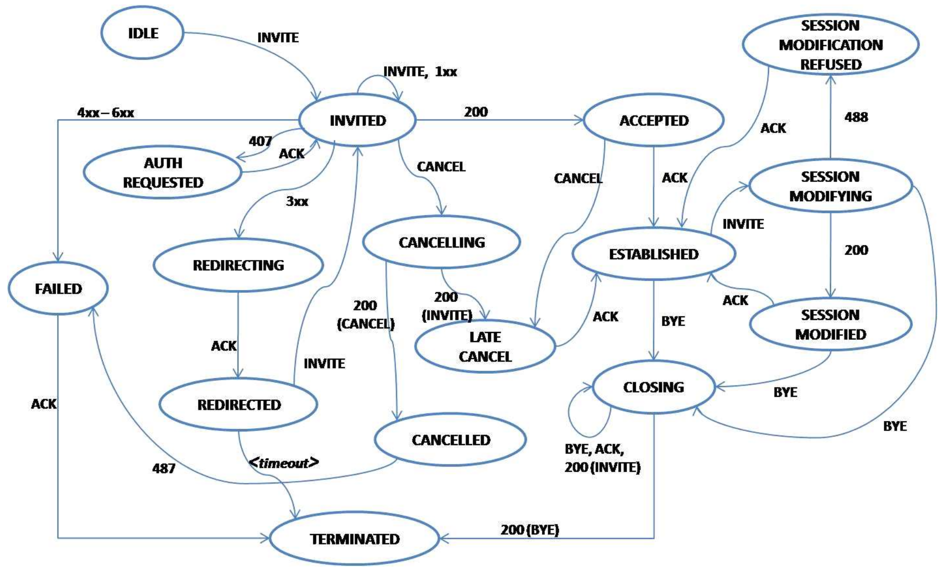


Figure 6.2: SIP finite state machine

ports 5060 and 5061, as well as on the media ports indicated by the SDP's *m-lines*. In Fig. 6.2, instead, we show the SIP Finite State Machine (FSM) we devised for the session module. We note that the detection process is based on the observation of packets flowing through a host's network interface, so it is a bit different from the classical SIP state machine.

The creation of a new SIP session is triggered by a new `INVITE` message and, within a SIP session, one or more RTP sessions might be created, each one representing a single medium. Specifically, the creation of an RTP session starts with the first SIP message that carries an SDP body (that could be either an `INVITE` or a `200`) and is completed as soon as the second SDP-carrying message is seen (a `200` or an `ACK`, respectively). An RTP session could also be created or modified by re-`INVITE` messages; we took into account such possibility since it is of key importance when both parties of the call make use of the ICE protocol. When the ICE negotiation ends, in fact, the caller sends a re-`INVITE` to update the media-specific IP address and port.

6.3.3 Case study: one-way media issue

As a case study, we show how we address the well-known issue of one-way RTP flows in VoIP communications by leveraging the DYSWIS diagnosing capabilities. We investigate the main causes that usually lead to this type of fault, and we detail the proposed methodology allowing for their automated online detection and diagnosis. As most of the problems associated with one-way RTP can be ascribed to the presence of NAT elements along the communication path, one of the key features of the proposed methodology resides in the capability to detect such type of devices. Besides, another important aspect of this work is that the diagnosis is non-intrusive, meaning that the whole process is based on the passive observation of flowing packets, and on silent active probing that is transparent to the users. In this way, we also avoid the possibility of being classified as SPIT (SPam over Internet Telephony). We provide a thorough description of the various steps the diagnosing process goes through, together with some implementation details as well as the results of the validation process.

Causes

The problem of one-way RTP flows is very common in VoIP communications. In this paragraph, we provide a classification of the causes that lead to such kind of fault, by splitting them into four main categories.

Configuration problems Into this category fall all the problems that can be ascribed to some error in the configuration of the machine hosting a User Agent (UA). First of all, there are possible oversights in the configuration of the UA itself (e.g., wrong audio capture device selected). Then, we have network interface configuration errors, that are quite common especially in multi-homed systems. In fact, it can happen to see RTP packets being received and sent on two different network interfaces, for example on machines having both a wired and wireless connection up (this is not unlikely on Unix-based systems, and is usually due to the configuration stored in the

`/etc/hosts` file). The presence of software firewalls not properly configured can also cause one-way media flows: for example, if we want both audio and video to be involved in the call, it would not be sufficient to open a couple of ports, since each call leg consumes two ports (one for RTP and the other for RTCP). Finally, we also classify IP address conflicts in the network as a local configuration problem.

As we will see in Section 6.3.3, it is easy to diagnose problems falling into this category.

NAT-related problems Most of the factors that can cause one-way media flows fall into this category and are related to the presence of NAT elements along the communication path. Several NAT traversal solutions have been proposed by the Internet Engineering Task Force (IETF), namely the STUN *(Session Traversal Utilities for NAT)* [14], TURN *(Traversal Using Relay NAT)* [15] and ICE *(Interactive Connectivity Establishment)* [28] protocols and the *Application Level Gateway* (ALG) and *RTP proxy* elements. If no such solution is employed, the User Agent is unable to receive RTP packets. Even worse, even if a NAT traversal technique is employed, it can happen that the "natted" party is anyhow unable to see incoming packets. This is the case of the most widespread NAT traversal solution: the STUN protocol. STUN is actually helpful in a number of cases; though, it is useless when a User Agent is behind a *symmetric NAT*[2], in which case it experiences one-way media flows. Furthermore, one more scenario where the STUN usage does not avoid one-way RTP flows is when both the caller and the callee happen to be in the same subnet, since a lot of NAT elements discard packets received from the private network and destined to their own public IP address. The last situation can happen also if the STUN protocol is not employed, but the NAT box has built-in SIP Application Level Gateway (ALG) functionality. This is becoming very common, as many of today's commercial routers implement such feature. Unfortunately, poorly implemented ALGs

[2]For a thorough description of the different types of NAT, the reader can refer to [14].

are quite common, too, and in some cases they can be the cause of the problem rather than the solution. Finally, very often the same device handles both NAT and firewall functions; in these cases, port blocking issues have to be taken into account.

Node crash problems The sudden crash of a network node also causes the inability to receive RTP packets. We remark that the crashed node could be neither the caller nor the called party, but a possible RTP proxy that belongs to the media path.

Codec mismatch A lot of SIP clients offer the possibility to select only a subset of media codecs, among the ones supported. Unfortunately, sometimes this choice is not reflected in the capabilities offered in the SDP, so it can happen that the result of the media negotiation is a codec that has been disabled. As a consequence of this, one of the parties involved in the call would not hear the voice or see the video of the other, even if it is actually receiving the corresponding RTP packets. We report this kind of problem just for the sake of completeness, as in this case we are not experiencing one-way media flows since RTP packets flow in both directions. Consequently, our work does not address this issue.

Diagnosis flow

Our goal was to diagnose one-way RTP faults by identifying the source of the problem among the ones presented above. We represent the whole process by means of a flow chart (see Fig. 6.3) that applies to both UAC and UAS scenarios. It takes into account all the scenarios that can lead to one-way media flows and, even if we will not thoroughly analyze all the possible branches, we provide, in Section 6.3.3, some reference scenarios that will help the reader understanding our work. In the diagram, the "local" adjective is used to identify elements or functionality that belong to the same subnet of the DYSWIS node which experienced the fault, while "remote" elements or functionality belong to the same subnet of the other party. We also make a distinction between *tests* and *probes*: the former class only exploits local information, while the latter plays an active role by introducing packets into the network. Finally, we explicitly mark the probes that need the help of a cooperating node in order to be performed.

We observe that it is not always possible to exactly identify the cause of the problem. The capability of making an accurate diagnosis, in fact, strictly depends on the complexity of the network topology under consideration and on actual availability of "remote" DYSWIS nodes, too. The ability to identify such nodes is of key importance and is far from trivial. In fact, when a remote node belongs to a private network environment (i.e., the remote party of the call is natted), its IP address is not helpful for our purpose. Even the node's *reflexive address* can be not helpful in cases where hierarchies of NATs are involved, like the one depicted in Fig. 6.4. We will explain in the following subsection how we coped with this issue.

It is worth remarking that one of our goals was to carry out diagnosis in a non-intrusive way. In other words, we did not want to allocate new "real" SIP call towards the caller or the callee, because they would be annoying and could be easily classified as SPIT. Instead, a DYSWIS node tries to collect as much information as possible: (i) from the observation of flowing packets, and (ii) with silent active probes (e.g., a STUN transaction to determine its own reflexive address). When an actual SIP session needs to be set up for diagnosing purposes, it is established between two DYSWIS nodes without using the default SIP ports, so that possible softphones running on those machines would not be alerted.

Description of tests and probes

In this subsection we provide a thorough description of the probing functions we designed and implemented. These probes allow us to test the network environments close to either the caller or the callee (e.g., NATs, ALGs), as well as possible external nodes, like RTP proxies.

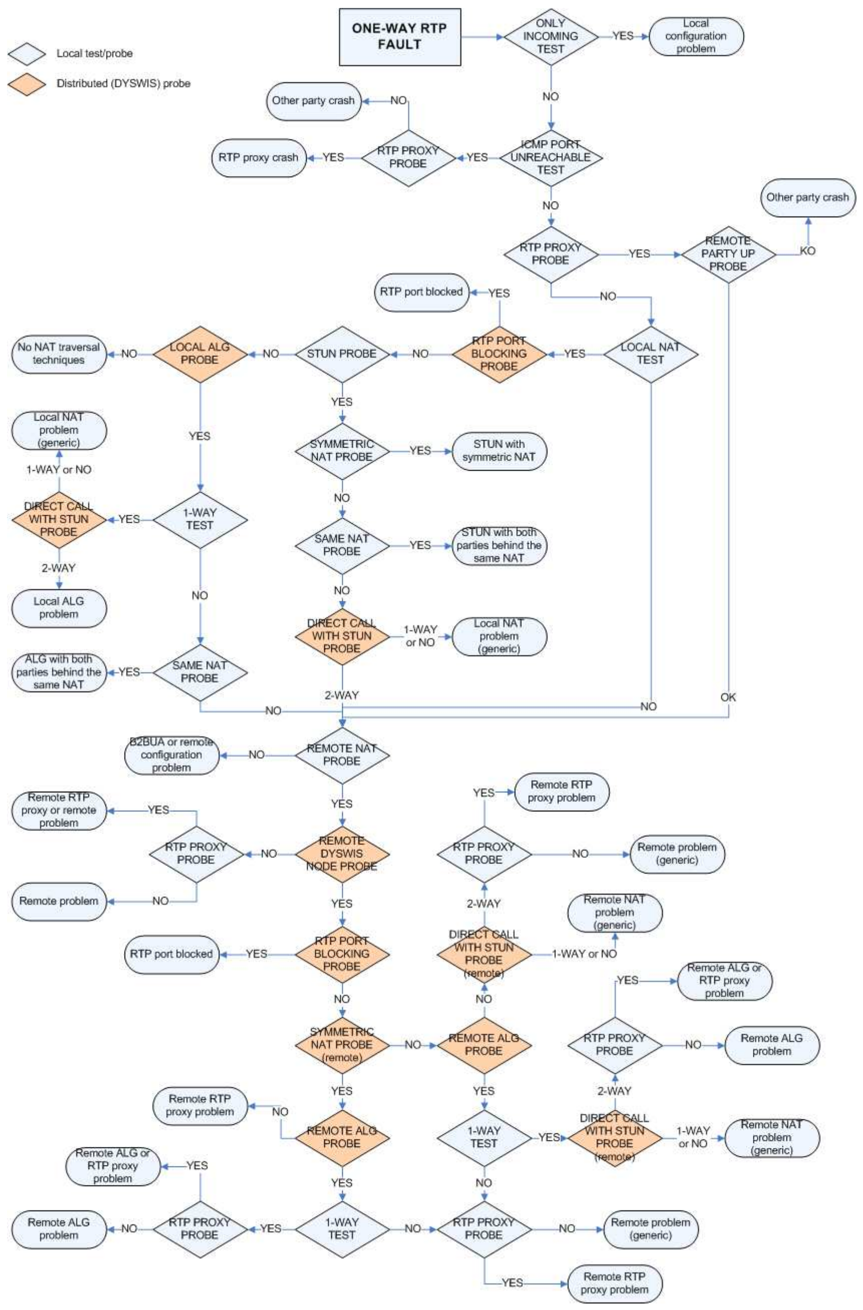


Figure 6.3: Flow diagram representing the whole diagnosis process

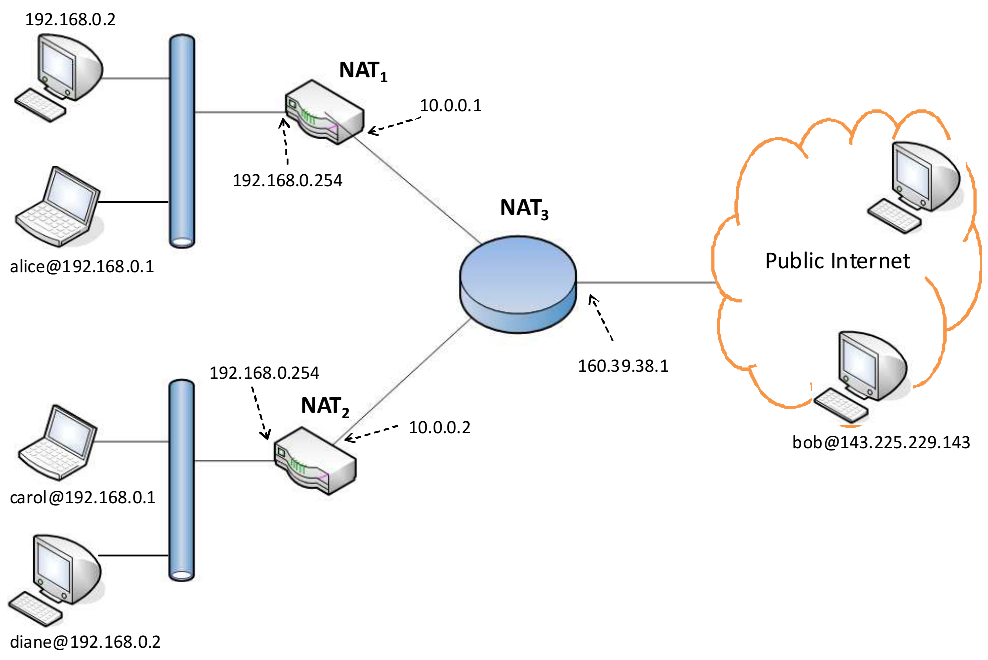


Figure 6.4: An example of NAT hierarchy that complicates the identification of "remote" peers

Only incoming test This is an easy test that checks whether the detected one-way RTP flow is only incoming or only outgoing.

ICMP port unreachable test Here, we check if there are incoming *ICMP port unreachable* packets, which would be a clear symptom that the process that was supposed to receive data is not active. Herein, we refer to this situation as a node crash.

RTP proxy probe This probe determines if there is an RTP proxy along the media path. An RTP proxy could be manually configured in the SIP client (e.g., a TURN server) or its usage might have been forced by a SIP proxy by modifying the SDP payload of the messages it forwards. We take into consideration both cases. For the former, we compare the IP address contained in the `Contact` header of an **incoming** message with the SDP's *c-line* **of the same message**: if they are different, we can presume that there is an RTP proxy. As to the latter case, instead, we inspect **outgoing** SIP packets, checking if the IP address contained in the SDP's *c-line* is different from both the local interface address and the reflexive IP address that is retrieved by means of a STUN transaction.

Remote party up probe Whenever an RTP proxy is employed, we are not capable to detect a possible crash of the remote node, since we would not receive any ICMP packet. In these cases, we check the availability of the remote party by sending a SIP `OPTIONS` message to it. Such message is sent through all the SIP proxies included in the signaling path, if any, in order to cross a possible remote NAT, making use of the `Record-route` and `Route` SIP headers.

Local NAT test This test determines if the local node (i.e., the node which experienced the fault) is behind a NAT by checking if the local interface has a private IP address.

RTP port blocking This probe verifies that the port number used for the RTP flow is not being blocked by a possible firewall running on the NAT box.

STUN probe Here we determine if the local node is making use of the STUN protocol. This probe consists in a STUN transaction to learn the local reflexive IP address. The result is then checked against the address contained in the SDP's *c-line* of an **outgoing** SIP message.

Local/Remote ALG This probe consists of a direct call attempt to a public DYSWIS node (i.e., a DYSWIS node that has a public IP address). As long as this call attempt is performed without exploiting any NAT-traversal technique, as well as without the SIP extension for Symmetric Response Routing [29], it lets us detect if the local or remote NAT has built-in Application Level Gateway functionality. In fact, the call attempt would succeed only if the private IP address, inserted by the client in the SIP message, is being modified by the NAT element before forwarding it. As previously said, we do not make use of the standard SIP ports for this call.

Direct call with STUN This probe differs from the previously described one only because the call attempt employs the STUN protocol.

Same NAT probe The public (reflexive) IP of the remote party is compared with the local reflexive address: if they match, the two parties are assumed to be behind the same NAT.

Symmetric NAT probe One functionality offered by the STUN protocol is the possibility to discover which type of NAT (Full Cone, Restricted Cone, Port Restricted Cone or Symmetric) is deployed. We use such feature to determine if there is a symmetric NAT, that, as already introduced, might be the cause of the fault we are trying to diagnose.

Remote NAT probe One of the main issues we had to face is the detection of remote NAT elements. In other words, we wanted to learn if the remote party is in a private network environment. Sometimes this is easy because, parsing a received SIP message, we find a private IP address (e.g., it could be in the SIP `Contact`, `From` or `To` headers, or in the SDP's *c-line* or *o-line*). Unfortunately, this depends on the specific implementation of the SIP element: for instance, some clients, when using STUN, put their public address in the SDP's *o-line*, while others do not. Similarly, some ALGs just parse outgoing messages and substitute every occurrence of a private IP, while others perform better thought-out replacements. When we cannot find any occurrence of private IP, we exploit a modified version of the IP traceroute we developed on our own, that sends a SIP `OPTIONS` message gradually increasing the *IP Time-To-Live* value. We send such request towards the public IP address of the remote node and, if we get an ICMP *TTL exceeded* packet whose source address is the original target of our request, it is a clear indication of the presence of a remote NAT element.

Otherwise, we could either receive a SIP response (e.g., a `200`) or do not receive any response at all. In the latter case, after having retried to send the message, with the same TTL value, for a couple of times (to take care of possible packet losses), we infer that there is a remote NAT box that is not a *Full Cone*. Consequently, our SIP message is being filtered. Finally, if we receive a response to the `OPTIONS` query, we cannot state there is no NAT along the path, yet. In fact, in the standard specification [33], there is no constraint for a NAT element to decrease the TTL value while forwarding packets. This topic has been discussed a lot on the BEHAVE mailing list of the IETF, where both personal opinions and implementation reports were provided. It turned out that a NAT does not always decrease the TTL of packets received on the public interface, while, for diagnostic reasons, it always decreases it for packets generated in the private environment and forwarded outside. Then, in order to take into account this possibility, when we receive a response to the aforementioned SIP `OPTIONS` query, we check the TTL value of the IP packet and try to infer whether it comes from a end-host or it has been modified by a NAT. This check is performed by considering that host operating systems have distinctive values for the initial TTL. Then, if the packet did not go through a NAT, the received TTL value would be equal to one of such initial TTL values, decreased by the number of "hops" returned by the traceroute. Otherwise, we infer the presence of a NAT. Further details of these OS-specific TTL values can be found in [20].

Remote DYSWIS node probe We conclude the description of the probing functions by showing how we realized the selection of a DYSWIS node that belongs to the same subnet of the remote party of the call. As we already said, a selection merely based on the public IP address would not be sufficient whenever there is a hierarchy of NATs. Then, after having selected all the DYSWIS nodes characterized by the same public IP address as the remote party, by means of the criterion described at the beginning of Section 6.3.1, we need to verify if one (or more) of them can be exploited for our purposes. We achieve this goal by sending a SIP `INFO` message in broadcast over the LAN. Such `INFO` message has to be sent within the dialog existing between caller and callee, so that, according to the INFO's RFC [11], *"A 481 Call Leg/Transaction Does Not Exist message MUST be sent by a UAS if the INFO request does not match any existing call leg"*. This is achieved by making the node aware of the `To` and `From` tags and of the `Call-ID`, so that it could be able to generate a request within a specific dialog. Therefore, each selected node would receive a non-481 response only if the remote party belongs to its same subnet.

Among all the methods envisaged by the SIP protocol, the only two that MUST[3] send an error response whenever they do not find any existing call leg are `INFO` and `UPDATE`. We chose to exploit the first one because, even if it is not mandatory, it is widely implemented in almost all the clients currently available.

Validation

In this section we provide the results of our validation. We tested our work with several different SIP clients. Specifically, we exploited the following softphones: *X-Lite* (Windows), *SJPhone* (Windows and Linux), *Ekiga* (Linux) and *PJSIP-UA* (Linux). As SIP and RTP proxies, we used *OpenSIPS* and its *RTPproxy* component, respectively. Finally, we developed our own implementation of a basic SIP ALG, since we could not find any suitable open-source library. With all these components, we set up a distributed testbed between the IRT lab at Columbia University and the COMICS lab at the University of Napoli. For the sake of conciseness, we do not present all the possible diagnosis paths that result from the flow chart in Fig. 6.3, which nonetheless have all been tested. Instead, we just provide a couple of representative scenarios, which show how the diagnosis process takes place.

[3]In the IETF jargon, the capitalized word "MUST" represents an absolute requirement of the specification.

Scenario 1: problem with the local ALG The first scenario we examine is characterized by the use of an ALG in the local network. We deliberately modified our ALG library in order to induce the one-way RTP fault. Specifically, we let our ALG function modify the *c-line* in the session-level section of the SDP message, without changing the same parameter in the media description section. So, since the session-level parameter is overridden by an analogous one in the media description, if present, the remote party will send its RTP packets to a private, non-routable, IP address.

In Fig. 6.5 we show a snippet of the whole flow diagram that applies to this situation, whose understanding is quite straightforward. We just clarify the last steps. The call attempted by the *Local ALG probe* can take place, thus revealing the presence of an ALG. Though, the resulting RTP flow is still one-way and this definitely represents a clue that the source of the problem might be the ALG itself. Such conjecture is confirmed by the *Direct call with STUN probe*. In fact, as long as we employ the STUN protocol before placing the call, the ALG does not come into play, since there would be no private IP addresses to replace.

Scenario 2: remote RTP proxy crash In this scenario, we suppose that both caller and callee use an RTP proxy. If the proxy used by the remote party crashes, the local DYSWIS node will experience a one-way RTP fault. Furthermore, it will not see any incoming ICMP packet (see Fig. 6.6).

In Fig. 6.7 we show the diagnosis steps in this scenario. We are supposing that the remote node is behind a non-symmetric NAT that has no built-in ALG functionality. However, even changing such hypotheses, we are still able to identify the cause of the fault. In general, when the diagnosis process involves the remote subnet, the results of the various probing functions allow us to narrow down the set of possible sources of the problem. In this case, we first get ensured that the problem cannot be ascribed to a remote ALG; then, we exclude that it could be somehow related to the remote NAT's behavior, since the SIP+STUN call involves two-way media flows. This brings us to

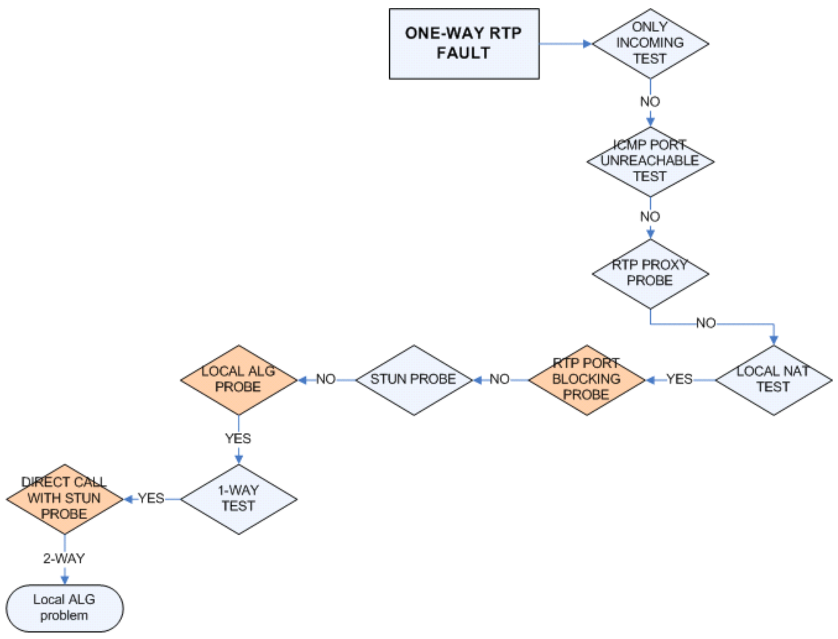


Figure 6.5: Local ALG problem

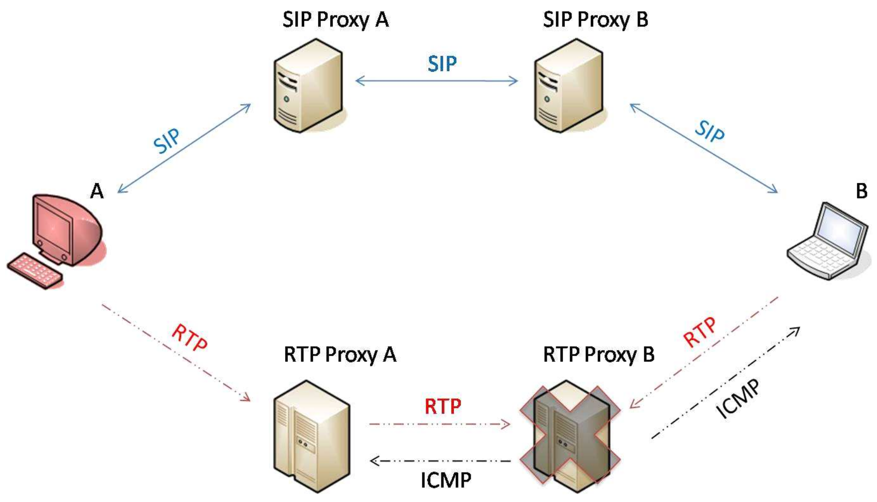


Figure 6.6: Remote RTP proxy crash: network topology

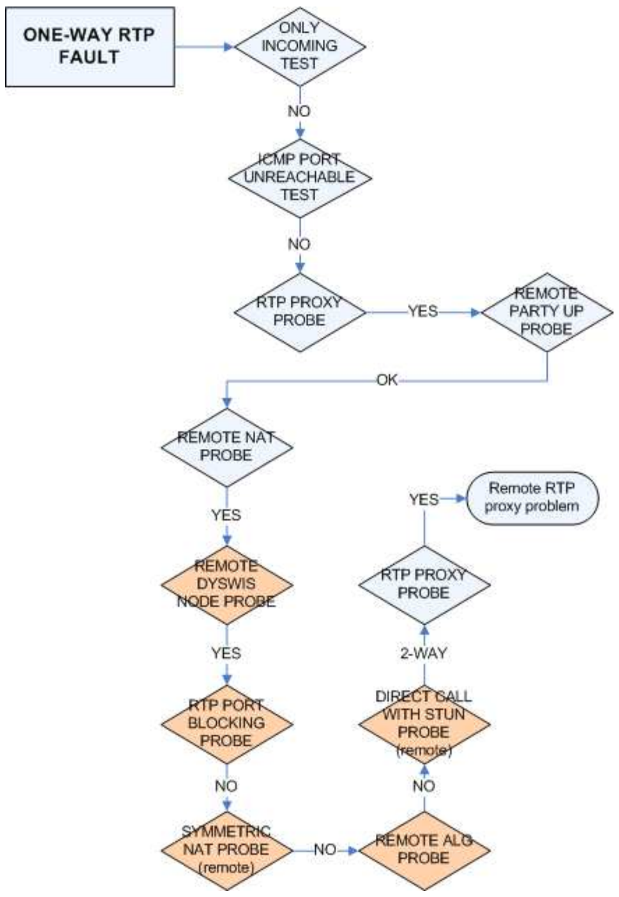


Figure 6.7: Remote RTP proxy crash: diagnosis flow

the final verdict. We observe that, in this lucky case, we are able to detect the exact cause of the fault, while in other cases, when the network topology is particularly complex, we are able to narrow down the fault space to two possible choices.

6.3.4 Implementation details

In this section we provide some brief information about the implementation choices. Besides Java, that has been chosen at the outset as the programming language for the whole framework for its well known platform-independence characteristic, the framework exploits the Jess rule engine[4] to control the diagnosis process. Jess uses an enhanced version of the *Rete* algorithm [12] to process rules, making Java software capable to "reason" using knowledge supplied in the form of declarative rules. Consequently, we implemented the whole flow chart presented in Fig. 6.3 as a set of rules in the Jess scripting language. The example below shows the rules allowing for the detection of a node's crash, when incoming ICMP packets are detected:

```
(defrule MAIN::RTP_ONEWAY
 (declare (auto-focus TRUE)) => (rtp_oneway (fetch FAULT))
)

(deffunction rtp_oneway (?args)
 "one-way RTP diagnosis"

(bind ?result (LocalProbe "RtpOnlyIncomingTest" ?args))(
  if (eq ?result "ok") then
    (bind ?finalresponse "Local configuration problem")
  else then
    (bind ?result (LocalProbe "IcmpDestUnreachTest" ?args))(
    if (eq ?result "ok") then
      (bind ?result (LocalProbe "RtpProxyTest" ?args))(
      if (eq ?result "ok") then
        (bind ?finalresponse "RTP proxy crash")
      else then
        (bind ?finalresponse "Other party crash")
    )
    else then
      ...
```

As to the SIP/SDP functionality, we adopted the JAIN APIs[5] developed by the National Institute of Standards and Technology (NIST).

For the invocation of remote probes on nodes that happen to be in natted environments, we chose to make use of the *udp-invoker* library[6], slightly modifying it in order to fit our needs. More precisely, a remote natted node is contacted by means of a relay agent, as shown in Fig. 6.8: as soon as a DYSWIS node belonging to a private environment becomes available, it sends a *udp-invoker ping* message to the relay agent, which in turn stores the related public IP address and port. Such message is sent periodically, in order to properly refresh the binding in the NAT table. Then, if the probing

[4]http://www.jessrules.com/
[5]https://jain-sip.dev.java.net/
[6]http://code.google.com/p/udp-invoker/

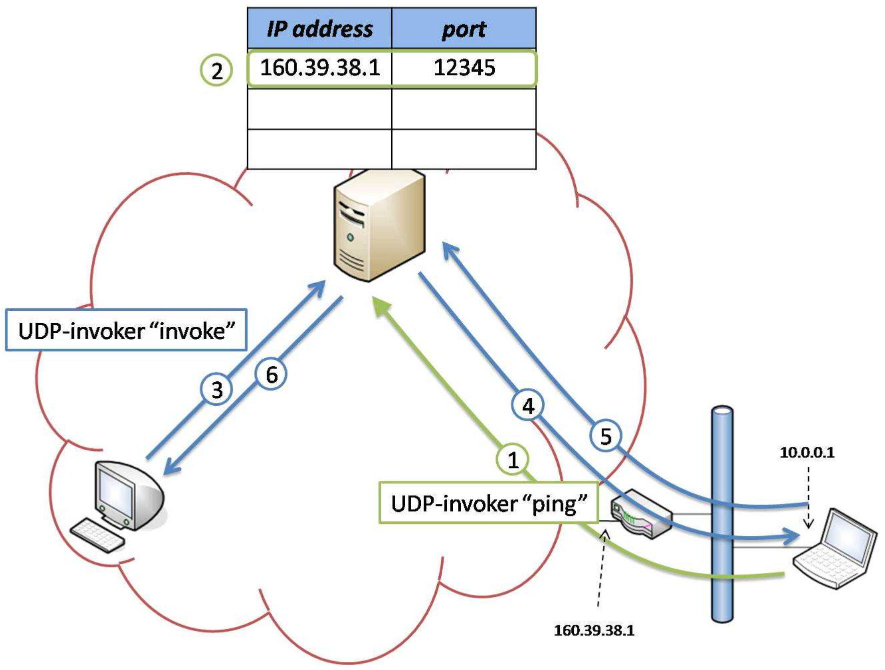


Figure 6.8: Remote probing functionality of natted nodes leveraging a relay agent

functionality provided by a private node needs to be exploited, the *invoke* message is sent through the relay agent. We remark that, in such way, we managed to cross any type of NATs. On the other hand, when the peer has a public IP address, the XML-RPC protocol[7] is exploited. Since it uses HTTP as the transport mechanism, it is more reliable than udp-invoker and, in some cases, it helps crossing restrictive local NATs.

Finally, the Jpcap library[8] allowed us to "sniff" packets from the network interfaces and send ad-hoc formatted packets, as well.

6.4 Meetecho tunneling solution

The previous chapters presented the reader with an overview on how multimedia conferencing has evolved in the latest year, by also mentioning the complex interactions among the numerous heterogeneous protocols it in-

[7]`http://www.xmlrpc.com/`
[8]`http://netresearch.ics.uci.edu/kfujii/jpcap/doc/`

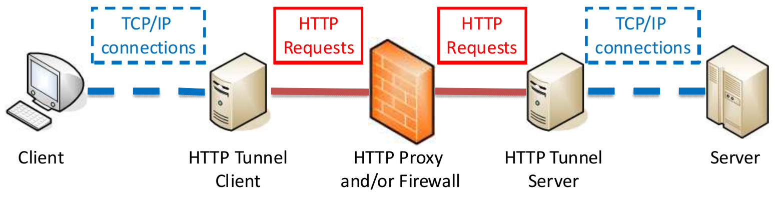


Figure 6.9: Generic HTTP tunnel

volves. Nevertheless, such protocols, despite being standards devised within the context of international bodies like the IETF, often fail to work when placed in real-world scenarios. In fact, as anticipated before, it is not unusual for a user to belong to restrictive network environments where the communication with the public Internet is regulated by proxy elements which only allow HTTP and/or HTTPS traffic to pass through. Solutions to effectively deal with them have already been proposed and exploited with alternating success during the years. Specifically, the most common approach that is exploited to get around such limitations is HTTP tunneling. This approach basically consists in encapsulating a generic protocol (e.g., SSH) in valid HTTP requests in order to allow such a protocol to travel across a network it would not otherwise be able to work in. This is usually achieved by deploying two entities into the network: (i) a tunneling server (acting as a web server) on a publicly reachable machine, and (ii) a tunneling client (acting as a web client) usually co-located with the protocol client which needs to be tunneled. A diagram explaining how HTTP tunneling works is presented in Fig. 6.9.

As it can be seen in the figure, the HTTP tunnel client acts as a server for the generic protocol client, while the HTTP tunnel server acts as the actual client for the generic protocol server. All the traffic is encapsulated by both elements in legitimate HTTP requests, and thus succeeds in traversing the installed HTTP Proxy/Firewall.

This approach is widely used; though, while it works fine for generic protocols like SSH or VNC, it raises issues when exploited to carry more sensitive protocols like the ones involved in multimedia conferencing scenarios. In fact, conferencing platforms often envisage the usage of several complex protocols to achieve the functionality they are designed for. Meetecho, for instance, makes use of many different standard protocols, namely CCMP for conference control, XMPP for instant messaging and notifications, SIP/SDP for negotiating multimedia sessions, RTP to transport audio and video frames in real-time, BFCP for moderation, specific protocols based on Java serialization features for what concerns white-boarding and desktop sharing, HTTP for presentation sharing.

That said, the first issue that comes to mind is related to transport-aware protocols like SIP and XMPP. In fact, as highlighted in Fig. 6.9, a tunnel breaks the end-to-end communication between the client and the server: the client sees the HTTP tunnel client as the actual server, while the server sees the HTTP tunnel server as the actual client. This means that, whenever transport-related information about the actual server and/or client is carried within the tunneled protocol, things are likely not to work at all. For instance, a SIP/SDP session would negotiate the wrong IPs/ports associated with media channels, and as a consequence no RTP connection would be established.

Besides, HTTP tunneling raises another issue related to real-time protocols like RTP. In fact, RTP usually is transported over UDP, considering it preferable for media packets to arrive soon rather than reliably. Nevertheless, HTTP is a TCP-based protocol, which means that encapsulated RTP packets may suffer from delay when transported to their destination. This is typically true whenever congestion control is involved.

Finally, considering almost all the protocols typically exploited within the context of multimedia conferencing sessions are bi-directional and asynchronous, ways have to be found to achieve a similar behavior on top of HTTP as well, which by itself is instead a stateless, request/response protocol.

From an architectural viewpoint, the tunneling solution we devised for Meetecho does not differ much from the typical HTTP tunneling solutions.

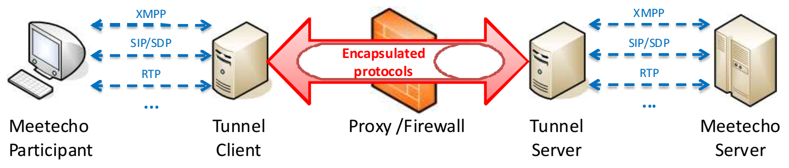


Figure 6.10: Meetecho HTTP tunnel

In fact, our approach is based on a client-server architecture, where the client and the server communicate with each other and take care of handling both the HTTP-based transport and the multiplexing/demultiplexing of the Meetecho protocols. An example is presented in Fig. 6.10. While the client is assumed to be co-located with the generic Meetecho participant (i.e. the client of our conferencing architecture), the server needs to be placed somewhere publicly reachable, in order to act as a web server that all clients, no matter how restrictive their reference proxy or firewall is, can reach via HTTP/HTTPS. As anticipated when describing the generic tunneling solutions, the Meetecho Participant assumes the tunnel client is the Meetecho Conferencing Server, while in turn the Meetecho Conferencing Server must think that the exploited tunnel server is the Participant itself. This means that the Participant does not login at the actual server, but at the tunnel client acting as a proxy for all the protocols involved in a multimedia conference. As a consequence, the Participant does not need to be aware of any tunneling being exploited: it can just assume the server is on a local address (i.e., the tunnel client), and the framework will do the rest.

To make this work, it is quite obvious that protocol rewriting may be involved. In fact, both the Tunnel Client and Server act as proxies/bridges for all the protocols. Since, as we already anticipated, some of these protocols are sensitive to transport-related information, proper care must be devoted to them, since a blind encapsulation would likely cause them to fail.

All protocols are encapsulated in one virtual channel: proper multiplexing and de-multiplexing is achieved by means of an ad-hoc protocol we devised ourselves. This protocol also takes care of additional functionality that may

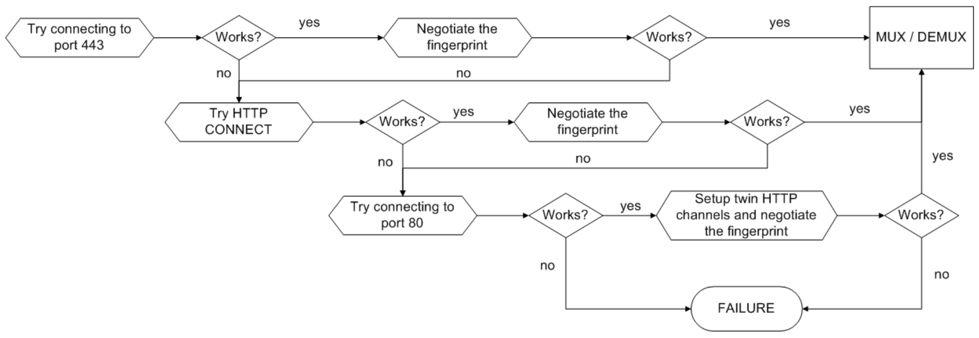

Try connecting to port 443
Works?
yes
no
Negotiate the fingerprint
Works?
yes
no
MUX / DEMUX
Try HTTP CONNECT
Works?
yes
no
Negotiate the fingerprint
Works?
yes
no
yes
Try connecting to port 80
Works?
yes
no
Setup twin HTTP channels and negotiate the fingerprint
Works?
no
FAILURE

Figure 6.11: Meetecho HTTP tunnel attempts: Flow Diagram

be required, like tunneling authentication, bandwidth monitoring/shaping of the managed sub-protocols, keep-alive mechanisms to avoid the underlying HTTP transport channel to be shut down, and so on.

In the following two subsections we will describe the HTTP transport approaches and how we encapsulated all the involved protocols, respectively.

6.4.1 Transport

As anticipated, we assume a single virtual channel is used to encapsulate all the protocols. Hence, we need a way to transport such a virtual channel over HTTP. In our architecture, we devised three different transport modes: (i) HTTPS, (ii) HTTP CONNECT, and (iii) HTTP encapsulation. The actual mode to be used changes according to the limitations enforced on the available network infrastructure, as it will be explained in the following subsections. A diagram depicting the sequence of encapsulation attempts is presented in Fig. 6.11

HTTPS

The first mode is HTTPS. The tunnel client tries to connect to the tunnel server via the well-known port number 443. In this mode, a single TLS connection is set up between the tunnel client and the tunnel server, since TLS encryption allows for the actual content of the connection to be hidden from the active proxy/firewall. This allows our framework to avoid making use of actual HTTP, but rather exploit our multiplexing/demultiplexing protocol encrypted in TLS. Pre-shared fingerprints are used to avoid potential man-in-the-middle issues. In fact, it may happen that HTTPS is allowed to pass by the proxy, but with the proxy itself acting as a man-in-the-middle to enforce some kind of payload inspection in order to make sure that the content is actually HTTP. Such a scenario would be detected by our platform, considering the fingerprint matching process would fail.

Considering the protocol contents are hidden, by relying on this mode of operation we can use the same connection both in transmission and in reception. In fact, we do not need to take into account the nature of HTTP, which means this mode is implementing a 'fake' HTTPS connection.

HTTP CONNECT

In case the fake HTTPS approach fails (either because the 443 port is blocked, or because the fingerprints do not match), the tunnel client tries the second mode, namely HTTP CONNECT.

CONNECT is a legitimate HTTP command which allows for the conversion of an HTTP request connection to a completely transparent TCP/IP tunnel. An example is the following:

```
CONNECT realserver.example.net:443 HTTP/1.1
Host: proxy.example.com:80
```

by which a client requests the forwarding of the present HTTP connection (between the client and the proxy) to a different server at a different port.

This was originally needed in order to instantiate HTTPS connections through HTTP proxies which did not make use of encryption. Nevertheless, since then many proxies disabled, or at least limited, support for CONNECT, considering it raised several security concerns. In fact, in the absence of further constraints every application could use HTTP CONNECT to carry a completely different protocol than HTTPS. As a consequence, the proxies that still accept CONNECT requests usually only allow it when the requested address is for a well known service, e.g., port 80 or 443.

That said, that is exactly what the tunnel client tries to do in case direct HTTPS fails: it tries to request from the proxy the creation of a new, still fake, HTTPS connection towards the tunnel server. Should the request succeed, everything happens just as if direct HTTPS were exploited.

HTTP Encapsulation

In case both the previous modes failed for any reason, we devised a third mode for the transport of the virtual channel. This mode is actual HTTP encapsulation. By using this mode, the tunnel client tries to connect to the tunnel server at port 80. If successful, actual HTTP messages are exchanged between client and server, meaning that HTTP headers and body are properly constructed.

Of course, some measures are taken in order to avoid problems. First of all, all the requests include headers to disable caching, in order to avoid any proxy to provide the client or the server with obsolete content after a request. Furthermore, since the content to be transported is not HTML but binary content, the *Content-Type* header must be manipulated as well in order not to have the proxy drop the request. We chose to simulate the transmission of JPEG images. Finally, two separate, yet logically linked connections are actually used to implement the asynchronous behavior needed by the virtual channel over the actual HTTP interaction. Specifically, the client issues two parallel HTTP requests: (i) an HTTP GET, to receive messages from the server, and (ii) an HTTP POST, to send messages to the server. Random *Content-Length* headers are chosen, in order to simulate different requests. As soon as all the data corresponding to a specific length have been sent/received, a new POST/GET is sent to keep the connection alive.

This approach has the disadvantage of bringing more overhead, considering additional encoding on the data is needed (in our case, we chose a *Base64* encoding) and two separate connections need to be properly synchronized.

However, this solution is the one more likely to succeed in the most restrictive scenarios, as confirmed by our experimental tests.

6.4.2 Protocols Handling

Whatever mode is exploited, what is transported is the same virtual channel. This channel acts as a virtual container for all the involved protocols, which are multiplexed (and de-multiplexed accordingly) at the application level. We also devised a solution able to cope with the co-existence of more instances of the same protocol in the same virtual channel, as in the case, for instance, of RTP, for which we might have a first connection dedicated to the audio stream and a second one devoted to video.

As anticipated at the beginning of this section, just encapsulating these protocols onto the virtual channel may not be enough in some cases. In fact, while protocols like CCMP, BFCP, RTP and (partly) HTTP can be blindly forwarded through the virtual channel without hassle, the same cannot be said for other protocols that are aware of the underlying transport information, as in the case of XMPP, SIP/SDP and Java serialization-based protocols (used by the white-boarding and desktop sharing modules in Meetecho).

These protocols are sensitive to some transport-level parameters that might be carried as part of their payload and hence call for proper manipulation and rewriting in order to work in a scenario like the one introduced by tunneling, which clearly violates the end-to-end principle.

XMPP

XMPP is used in Meetecho for several purposes. In fact, despite being a widely spread instant messaging protocol with support for Multi User Chat (i.e., conference rooms), it is also easily extensible with additional features (like notifications, voting/polling, dynamic negotiation of features, etc.) that make it the perfect candidate for conferencing-related functionality.

That said, XMPP is partly aware of the network it is deployed on. Specifically, domains and IP addresses play a strong role in XMPP, since they are part of the `To` and `From` fields of all requests, responses and events. In order not to suffer from the tunneling encapsulation, the domain part of all these fields needs to be properly rewritten, since the tunnel client and the tunnel server act as entry and exit points for the XMPP communication with the participant and the conferencing server, respectively.

Besides, as we already said, in the Meetecho platform XMPP is also used to dynamically negotiate some of the additional functionality provided. It is the case, for instance, of presentation sharing and desktop sharing. These features, while implemented by means of other protocols, are actually negotiated within the context of XMPP requests. This means that these requests need to be manipulated as well, so that the tunnel client can deploy virtual servers to act as bridges for those protocols as well, in place of the unreachable real servers.

SIP/SDP

SIP and SDP are widely deployed protocols devoted to the negotiation of multimedia sessions. They are the most commonly used protocols in VoIP scenarios. For this reason, since the beginning they have been chosen as the perfect candidates for the multimedia functionality in a standard conferencing environment like Meetecho.

They both are strongly aware of transport information, and as such very sensitive to anything that could interfere with it (e.g., NAT issues, etc.). Transport-related information, in fact, can be found in many of the headers in SIP, and in the negotiation lines of SDP. For this reason, all these fields and lines need proper rewriting in order to take care of the protocol bridging.

For what concerns SIP, the REGISTER message needs additional care, considering the negotiated DIGEST includes the involved addresses.

For what concerns SDP, instead, all that is needed is a proper rewriting of the media lines, in order to create RTP bridges accordingly.

Java Serialization

Java Serialization is a technique commonly used in several network projects based on Java. It basically consists in a mechanism by which native Java objects can be converted in a binary format, for instance in order to be transmitted over a network stream, and converted back to native Java objects once they reach their destination. Such technique is used in our conferencing platform for two main functions: shared whiteboard (*jSummit*) and desktop sharing (*jrdesktop*).

By itself, Java Serialization is not a network protocol, and as such it is not sensitive to transport-related information. Nevertheless, protocols can be built on top of it, as is the case of Java Remote Method Invocation (Java RMI) and the aforementioned Meetecho features. Whenever such protocols include the manipulation of objects which contain addresses and/or ports in order to work correctly, they become sensitive to tunneling. This is exactly the case of both *jSummit* and *jrdesktop*.

In order to make them work despite the tunneling, we had to implement a proper manipulation of the protocol contents. This included low-level manipulation of the serialized objects in the streams, making sure they remained coherent and decodable at the destination (e.g., enlarging/restricting payloads when necessary in order to avoid exceptions).

6.4.3 Experimentations

This section presents a brief overview of the experimental results we obtained from the implementation of our approach. We both show how we verified the capability to deal with restrictive network environment, and report some performance figures we devised.

Proxy/firewall traversal

In order to simulate a restrictive component on the network, we deployed a generic PC equipped with a Linux distribution (Debian GNU/Linux 5.0) in our testbed, to act as either a firewall or a more or less restrictive proxy.

We made use of well known open source tools to implement these behaviors, namely IPtables for the firewall functionality, Apache HTTPD as a generic HTTP proxy and Squid as both an explicit and a transparent HTTP proxy.

To make sure all the traffic coming from the participant would need to pass through our component in order to reach the server, we configured the PC as a Linux Router, and then explicitly set it as the default gateway for our test participant.

For what concerns the firewalling functionality, we prepared a series of increasingly restrictive IPtables rules to enforce on such a machine, in order to filter the traffic handled by the proxy accordingly. This allowed us to force a test of all the three transport modes previously introduced, namely HTTPS, CONNECT and HTTP.

To test the first mode, the first rule we envisaged was to only allow HTTP and HTTPS traffic through the proxy. This, translated to IPtables words, means we needed to prepare a rule to only allow incoming TCP connection on the well-known ports used by those two protocols, i.e., 80 and 443, as shown below.

```
iptables -I FORWARD 1 -p tcp -m multiport --dports 80,443
  -j ACCEPT
iptables -I FORWARD 2 -m state --state ESTABLISHED,RELATED
  -j ACCEPT
iptables -I FORWARD 3 -j DROP
```

In this first setup, only IPtables was configured, while neither the Apache web server nor the Squid proxy were active.

We verified that, under these restrictions, the Tunnel Client was able to reach the Tunnel Server using the fake HTTPS operation mode, and allowed its related participant to successfully join a multimedia conference. In fact, as explained, the fake HTTPS has the Tunnel Client connect at the Tunnel Server at port 443, thus simulating a HTTPS connection and circumventing the IPtables rule. Of course, considering such an experiment only envisaged a blind port-filtering mechanism to be in place, it was assumed to work fine anyway. In fact, whenever ports alone are used to filter traffic, these rules are easily circumventable.

After this trivial experiment, we chose to enforce a stricter restriction on the allowed connections, in order to test the second mode as well. We first removed the 443 port from the previous rule, thus limiting the allowed traffic to HTTP only:

```
iptables -I FORWARD 1 -p tcp -m multiport --dports 80 -j ACCEPT
```

This time, nevertheless, we also deployed an Apache HTTPD Web Server in order to add support for the HTTP CONNECT to our proxy component. To do so, we added the following lines to the HTTPD configuration:

```
<IfModule mod_proxy.c>
    ProxyRequests On
    AllowCONNECT 443
    [..]
```

which allowed HTTPD to act as a proxy, but only for requests directed to a server at port 443, thus limiting the proxy functionality to HTTPS only. As expected, this time the fake HTTPS mode was not successful: in fact, the IPtables rule did forbid any connection attempt to any port different than 80, and so 443 made no difference. As a consequence, the Tunnel Client tried the next mode, i.e., HTTPS via CONNECT. Having explicitly configured the application to use our proxy component as HTTP proxy of the network, the Tunnel Client successfully managed to reach the Tunnel Server: in fact, first of all the request was addressed to a server listening on port 80 (HTTPD), thus circumventing the limitation imposed by IPtables. Besides, the request was an HTTP CONNECT request to reach the Tunnel Server at port 443: this was allowed by our restriction on the proxy functionality in the HTTPD configuration, and as a consequence the Tunneling worked fine as well.

Having verified that the Apache web server supported our request, we chose to also test the second mode using a more strict component as the Squid proxy. Squid is a well known open source component, and it is the most commonly deployed component whenever caching and proxying functionality are required for HTTP. That said, we disabled Apache and launched Squid as an explicit proxy instead. The support for CONNECT, as before, was limited for port 443 alone, which in Squid words is translated like that:

```
acl SSL_ports port 443
acl CONNECT method CONNECT
http_access deny CONNECT !SSL_ports
```

Even in this scenario, the second mode of operation was successful: the Tunnel Client was able to request the creation of an encrypted connection towards the Tunnel Server by means of the CONNECT method, and the session went on as expected. This confirmed us that the approach we employed was correct and effective for such a need.

To test the last and more delicate mode, we removed the support for the CONNECT method from our testbed scenario. Maintaining the previous IPtables rules enforcements, we also changed the default Squid behavior, by making it act as a transparent HTTP proxy rather than an explicit one: this means that we wanted all HTTP traffic to be intercepted and manipulated by Squid without any configuration needed on the client side, making all clients unaware of any proxy/cache being in place. In order to do so, we first configured Squid as a transparent proxy:

```
http_port 3128 transparent
always_direct allow all
```

Then, we also configured a proper IPtables rule to forward all requests addressed to port 80 to port 3128 instead (the default port used by Squid), in order to actually redirect all the HTTP traffic passing through our component to Squid to have it managed accordingly.

```
iptables -t nat -A PREROUTING -i eth1 -p tcp --dport 80 -j DNAT
  --to 192.168.1.1:3128
iptables -t nat -A PREROUTING -i eth0 -p tcp --dport 80
  -j REDIRECT --to-port 3128
```

We then launched the experiment again. As expected, the first two modes both failed: HTTPS on port 443 was blocked by IPtables, and no HTTPS proxying functionality was available to attempt an HTTP CONNECT bridged connection. The third mode, instead, worked flawlessly: in fact, such mode assumes two HTTP connections are set up towards the Tunnel Server at port 80, thus successfully traversing the IPtables limitations. Besides, Squid successfully recognized both the connections as valid HTTP traffic, and redirected the traffic to the Tunnel Server according to its business logic.

RTP over TCP performance assessment

The transmission of real-time media flows usually leverages the UDP protocol to transport RTP packets. In fact, a reliable transmission might be inappropriate for delay-sensitive data, since retransmissions increase delay and congestion control mechanisms affect the transmission rate. We presented, instead, a tunneling solution that exploits the HTTP protocol (which works over TCP connections) to carry RTP packets, too. In this subsection we present the results of the tests we performed in order to assess the performance of the proposed approach. We exploited our Meetecho conferencing platform to evaluate the delay that characterizes the RTP connections, either when no tunneling solution is in place or when the Meetecho tunneling solution is employed. More precisely, we focused on the delay jitter, which is known as the key performance indicator of real-time communication channels. Fig. 6.12-a shows the time evolution of the jitter, as regards an audio RTP channel relying on the UDP protocol. Fig. 6.12-b, instead, depicts the jitter evolution when tunneling is employed in an uncongested network environment, which is very similar to the previous one.

Finally, Fig. 6.13 shows how tunneling affects the performance when a heavy network congestion occurs: in this scenario the trend is more irregular, reflecting the further delay introduced by TCP congestion control mechanism. Anyway, from a qualitative perspective, such delay did not lead to

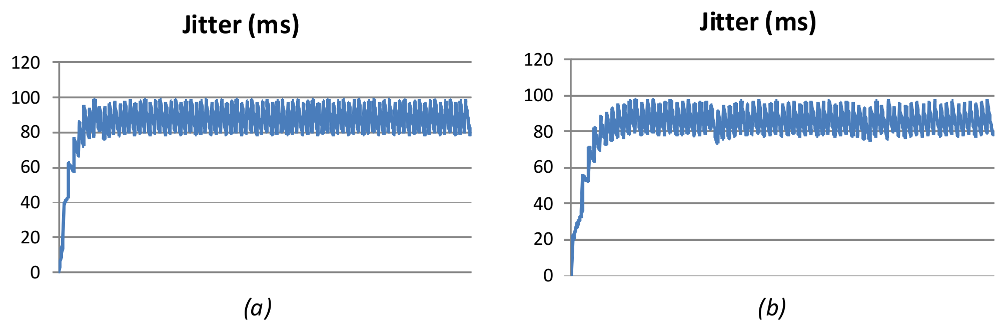


Figure 6.12: Time evolution of the jitter with no tunneling solution (a), and when tunneling is employed in an uncongested network (b).

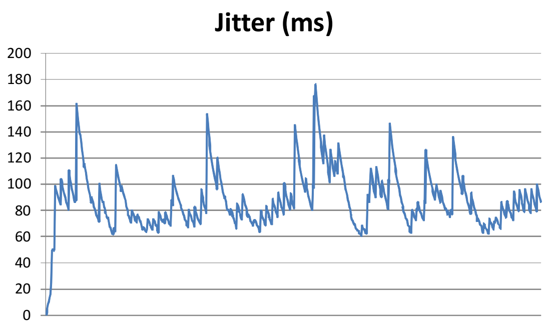


Figure 6.13: Time evolution of the jitter when tunneling is employed in a highly congested network.

bad quality audio communications, since the conference audio was still intelligible.

Chapter 7

Conclusions

The work carried out during the last three years as Ph.D. Student has been focused on the multimedia conferencing service over next-generation IP networks. The ultimate goal was to design an architecture having high scalability requirements, while taking into account the outcome of the standardization efforts conducted by the Internet Engineering Task Force.

Fig. 7.1 depicts a progressive workflow, with the three main milestones met: *centralized conferencing*, *distributed conferencing* and *troubleshooting.*

As to the former, we thoroughly examined the state of the art, pointing the attention on the standardization efforts the conferencing service was going through (see Chapter 1). Moreover, I actively contributed to the definition of the state of the art itself, by actively participating to the activities carried out within the XCON and MEDIACTRL Working Groups of the IETF. In such context, reference testbeds implementing the specified frameworks and protocols have been realized, as well as call flow documents which are on the path to become *Informational RFCs.* The output of the XCON-related work has been *Meetecho*, a standards-compliant multimedia conferencing architecture which also leverages the MEDIACTRL approach as a first step towards scalability (see Chapter 2 and Chapter 3).

After the work upon the Meetecho centralized platform has been completed, we started investigating a possible evolution of the XCON framework towards distributed environments. We came up with the DCON (Distributed

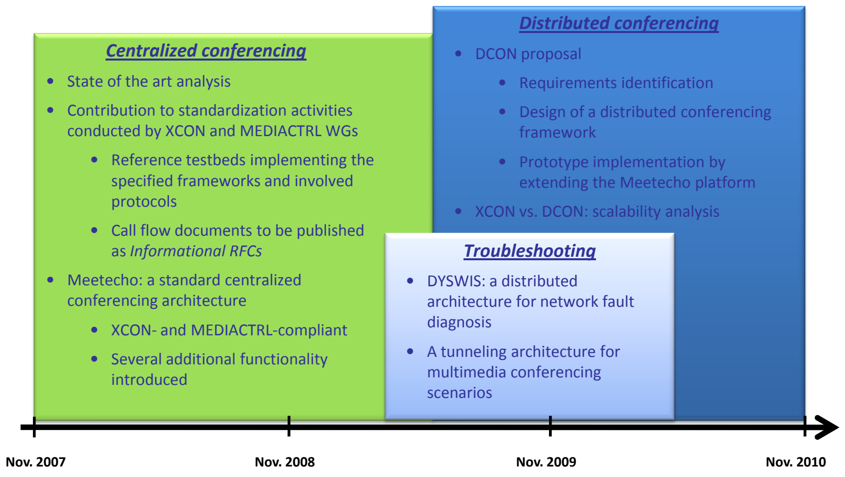

Centralized conferencing
State of the art analysis
Contribution to standardization activities conducted by XCON and MEDIACTRL WGs
Reference testbeds implementing the specified frameworks and involved protocols
Call flow documents to be published as Informational RFCs
Meetecho: a standard centralized conferencing architecture
XCON- and MEDIACTRL-compliant
Several additional functionality introduced
Distributed conferencing
DCON proposal
Requirements identification
Design of a distributed conferencing framework
Prototype implementation by extending the Meetecho platform
XCON vs. DCON: scalability analysis
Troubleshooting
DYSWIS: a distributed architecture for network fault diagnosis
A tunneling architecture for multimedia conferencing scenarios
Nov. 2007
Nov. 2008
Nov. 2009
Nov. 2010

Figure 7.1: Progressive workflow

Conferencing) proposal, which envisages a distributed architecture where a number of XCON-compliant "islands" are interconnected through an overlay network (see Chapter 4). DCON has been implemented extending the Meetecho platform, which we also used as a term of comparison while evaluating performance figures. The experimental campaign we conducted proved how the migration towards distributed environments definitely improves the scalability of the overall system (see Chapter 5).

Finally, the last milestone is related to troubleshooting. In fact, when we brought the results of our research activities from the lab to the real world, a number of issues arose. Such issues have been identified and solved by means of a network diagnosis architecture and a protocol tunneling solution, respectively (see Chapter 6).

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