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A Development and Runtime Platform for Teleconferencing Applications

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Abstract—**A communication platform is described that supports the fast implementation of networked multimedia applications with conference character and collaboration features. The platform exhibits the notion of a site as one of its main abstractions. A site is a collection of workstations, media input and output devices that are, in terms of control, tightly coupled. Connection and application control is centralized within one site, but distributed among different sites. The platform exports a programming interface with high-level abstractions for session and connection control, allowing application developers to concentrate on scenario and user interface design. The platform was implemented in the course of the European Beteus (Broadband Exchange for Trans-European Usage) project. A tele-meeting application and a teleteaching application were developed on top of it. Platform components and applications were tested on the European ATM pilot network over a period of nine months. The paper first describes platform architecture and programming interface; it then talks about the implementation of platform and applications, and their deployment in the harsh environment of a trans-national broadband pilot network.**

I. Introduction

Today's collaborative teleconferencing systems are usually implemented as stand-alone applications with fixed interaction and communication scenarios. They establish static audio and video connection structures among the conference participants and employ a specific tool for collaboration. The software architecture of such systems is often highly rigid; since it is designed with the requirements of a single application in mind it does not automatically support the reuse of its components within other application scenarios. This means that there is a new software design and implementation process each time a new application needs to be developed, with code reuse being at the library level or lower. It is clear that the standalone system approach to teleconferencing application development will give way to a platform approach. Networked multimedia applications in general will be more and more implemented on top of programming interfaces that provide different levels of control for media acquisition, transmission and playout, or simply for whatever building block is likely to be used by a large number of applications. Some of these interfaces will be standardized, allowing applications developed for one hardware architecture to be easily ported to other ones. Authoring tools and application development platforms will further ease the design and implementation of networked multimedia applications. An application development platform is especially necessary in the case where an application is to be offered as a service in a large public or private network. Much of the complexity there stems from the necessity to integrate the application into the network infrastructure and to make it interwork or coexist with other services. Development platforms do exist for the creation of interactive retrieval services on residential cable networks. As the customer access link becomes symmetric, there will be a demand for multi-point and multi-user services like tele-conferences, having life-cycles maybe just as short as those of retrieval services. Development platforms for multi-user services will have to deal with dynamic connection structures and with multiple user interfaces, just to name two sources of added complexity.

The platform approach may also be of advantage in areas other than service provision. The particular problem we were faced with was to develop a set of applications for the European Beteus (Broadband Exchange for Trans-European Usage) project [1]. The project definition of Beteus focussed on the network communication aspects rather than the applications. The main requirements on the applications were that one of them be a tele-teaching application, and that they make the best use of the high bandwidth available on the ATM network that interconnects the project partners in France (Eurécom in Sophia-Antipolis), Switzerland (CERN in Geneva, EPFL in Lausanne, ETHZ in Zürich) and Germany (TUB Berlin). Two applications were vaguely envisaged, a tele-meeting application for informal group meetings, and a distributed classroom application that would allow to give a lecture at one site to a virtual classroom that is the combination of classrooms at several Beteus sites. Since there was no clear vision for the applications at the beginning of the project, it was decided to build an application platform rather than stand-alone applications for everyone of the envisaged application scenarios. The platform should constitute the highest common denominator between the envisaged application scenarios and should allow to implement and to incrementally improve an application scenario with significantly reduced effort as compared to an approach based on stand-alone prototypes.

The Beteus platform and the initial application scenarios were designed and developed in the period from August 1994 to April 1995 [2]. In May 1995, the Beteus field trials started on the European ATM pilot network. Two application scenarios were successfully demonstrated to a commission of the European Union in July [3]. The field trials continued until the beginning of December 1995, with the second major event being the coorganization of a distributed conference on November 16 and 17 between a main site in Madeira and attached sites in Madrid, Brussels and Sophia-Antipolis (IDC‘95). The deployment of the platform on an experimental trans-national ATM network turned out to be an important, but not always easy experience.

The paper is roughly divided into an architecture part and an experience part. The first part starts off with a section about the design issues and constraints that underlie our main architectural decisions (Section II). It then gives an introduction into the Beteus application model and a complete description of platform architecture and components (Section III), followed by a summary of the main session and connection related abstractions (Section IV). Based on this, the application programming interface is presented. An extended example demonstrates its main features as well as the application development methodology (Section V). The beginning of the second part of the paper is marked by a section discussing implementation details of platform and applications (Section VI). The Beteus network configuration and the trials are described, with the main emphasis on how multicast is provided to applications (Section VII). Following a short discussion of related work (Section VIII), a final conclusion presents some ideas for the directions into which the Beteus platform can be developed in the future.

II. Objectives and Design Issues

Beteus is a follow-up to the Betel (Broadband Exchange over Trans-European Links) tele-teaching project in which two of the Beteus partners, Eurécom and EPFL, were engaged [4]. An overview of Betel is given in order to clarify the historical background of Beteus.

A. The Betel Project

Betel implemented a tele-tutoring application scenario in which a professor in Lausanne supervised the laboratory exercises of students in a classroom at Eurécom in Sophia-Antipolis. Both professors and students had personal workstations that were equipped with microphone, speaker, camera and an external video screen. There was additional equipment on classroom level: a large screen in front of the student workplaces, a camera that overlooked the classroom, a speaker and a microphone. A typical session started with the professor giving an introduction into the laboratory exercise. He was visible and audible at classroom level and saw and heard himself the whole class. The students then started to work with some application, in this case an intelligent network simulator, and the professor waited for questions to come up. A student who needed help pushed the question button in the user interface and waited for the professor to connect to him. The professor in turn would see the question on his interface and push the answer button, which would make him visible and audible at the student’s personal workplace. The student could then share the simulator application with the professor, and the professor could guide the student remotely to the solution of his problem.

In Betel, EPFL and Eurécom were connected with a 34 Mbit/s ATM link via a cross-connect in Lyon. ATM was not directly visible to applications; a Betel end-station was connected via a router to a network adapter that encapsulated Internet Protocol (IP) packets in Switched Multimegabit Data Service (SMDS) packets before transmitting them over the ATM link. At Eurécom, audio and video of the professor were distributed with an analog switch. People that attended the demonstrations were very much impressed by the dynamic changes of the connection structure in the classroom, i.e., by the ease with which the professor’s virtual presence moved from classroom level to particular students and back. The analog switch was hiding the fact that there was at no point more than one digital audio and video stream in one direction between Lausanne and Sophia-Antipolis.

Beteus should build on the experience gained with Betel. It should improve Betel in various respects: it should provide true digital multi-point communication and a minimum of two application scenarios. In addition to that, Beteus was to be employed for at least one event involving real users, as opposed to the theatrical demonstration of Betel. One proposition for this was to organize a distributed summer school for business students where the Beteus platform would be used for formal lectures, panels, group presentations and the like. It was clear that for such an event to be successful the platform would need to have a degree of maturity that allows non-skilled users to be let alone with it.

B. The Beteus ATM Network

At the beginning of the project it was not clear how exactly the project partners would be interconnected with each other. It was assumed that the majority of the project partners would have access to the European ATM pilot network, but at least in the case of Eurécom it looked a long time as if access would be SMDS as in Betel. It was a clear objective to have ATM access for all project partners since such an access was supposed to be favorable for multipoint communication. The network configuration that was finally reached is qualitatively depicted in Fig. 1. All Beteus project partners had ATM LANs. The ATM LANs of all partners but ETHZ connected to the ATM pilot via 34 Mbit/s E3 interfaces. ETHZ is the only project partner that had a 155 Mbit/s STM 1 link to the ATM pilot. The ATM pilot itself is a collection of ATM cross-connects in various European countries. Beteus ran over cross-connects in Paris, Cologne, Geneva and Zürich as far as can be judged from the scarce information provided by the network operators.

Although the resulting network was a pure ATM network it was never seriously considered to use any network protocol for the application platform other than the Internet Protocol (IP

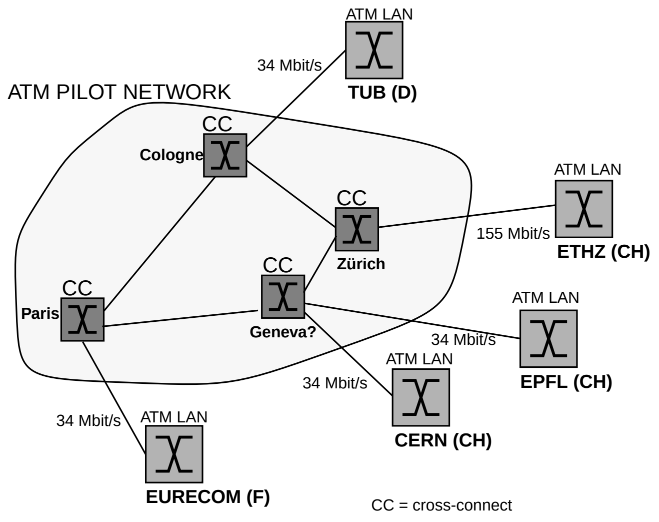


Fig. 1. The Beteus ATM network.

over ATM), and any network programming interface other than the Berkeley sockets. The use of proprietary ATM programming interfaces was not considered, first because this would have defeated platform portability, and then because of the performance problems that implementations of such interfaces still exhibit.

C. Application Scenarios

The broadband ATM network interconnecting the project partners was a key aspect of Beteus. The Beteus applications were supposed to demonstrate the high quality of human communication and interaction that can be achieved when bandwidth is not a limiting factor. The people that were brought together by a Beteus application should communicate and interact as freely as if they were sitting together around a table in a conference room [5]. This is only possible if the quality of the audiovisual communication and of the collaboration tools is such that geographically dispersed users are perceived as being present at every implicated location. In addition to that, a user must have the impression that he is seen and heard; he must trust the system to really convey his image and speech to other users.

With a possible summer school event in mind, two application scenarios were envisioned at the beginning, a tele-meeting scenario and a distributed classroom scenario. The tele-meeting scenario should be a rather informal meeting environment where people can come together for discussion and collaboration. The distributed classroom scenario should combine classrooms at different sites to a single virtual classroom. A professor can give a lecture in one classroom in which remote classrooms participate. Every classroom is equipped with multiple screens that show all other classrooms and possibly the slides of the professor, and people within different classrooms can communicate with each other and the professor in a way similar to a panel discussion. The two scenarios differ fundamentally from each other in that the first one assumes a single user terminal as standard endpoint equipment, whereas the second uses a collection of workstations and media input and output devices to assemble a classroom. None of the scenarios was clearly specified at the beginning of the project, and it was not even sure if both would be retained. It was assumed that the application scenarios would evolve in the course of the project as tests are performed and experience is gained.

These considerations led to the decision to implement a real conferencing platform rather than stand-alone systems for everyone of the envisaged application scenarios. The conferencing platform should offer a high-level application programming interface (API) with which the effort to implement an application scenario could be kept at a minimum, with this minimum being not much more than the effort to implement the graphical user interface (GUI). In addition to basic audio and video support the platform should also contain a collaboration tool. It was therefore decided to integrate a shared window system into the platform. It was also decided to integrate a light-weight directory service that provided a run-time framework for the application scenarios. Users would log formally into the platform; once logged they could consult the directory service to see who is there, and then start or join applications or just remain passive.

At a later stage of the project it turned out that the official event of the project would be a distributed conference, and not the summer school. Since the distributed conference could be handled with the tele-meeting the decision was taken to postpone the implementation of the distributed classroom scenario and to implement instead a third scenario, the tele-tutoring scenario, which is a replication of the Betel application on the Beteus platform.

D. Basic Software Architecture

An important design issue was the control architecture of the platform. In terms of control, the platform could be completely centralized or completely distributed. The centralized solution was declined, first because there would be a single point of failure, then because of performance and scalability considerations. But it was felt that control should be centralized within the network of a project partner. This is because it was assumed that an application endpoint will not be a single multimedia workstation, but rather a logical unit that is assembled from a collection of resources including workstations, multiple screens, cameras, speakers and microphones, digital and analog switches. A combination of equipment is found in the distributed-classroom scenario, but it is also interesting for the realization of personal workplaces like in the tele-meeting scenario where a single workstation processing multiple audio and video streams may easily run into performance problems. Some of this equipment, especially workstations and analog switches, would be shared by many logical application endpoints, making it necessary to have some central connection management entity. The scope of this central connection management entity is limited to the local network, with the exact composition of an application endpoint being hidden to the outside. The establishment of a connection between the networks of two project partners must therefore involve some communication between the respective connec-

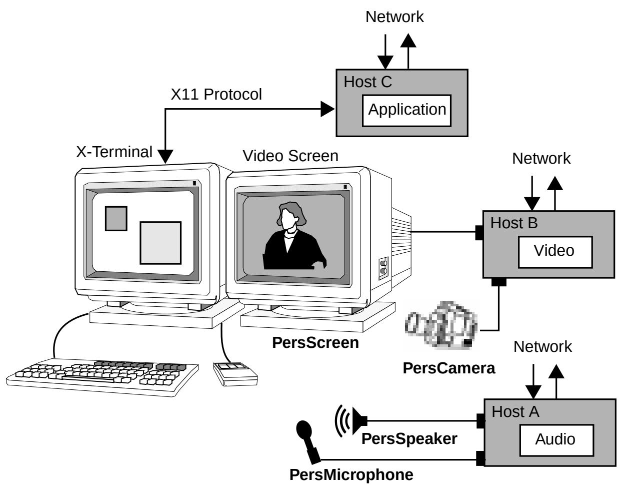


Fig. 2. Node mapping example

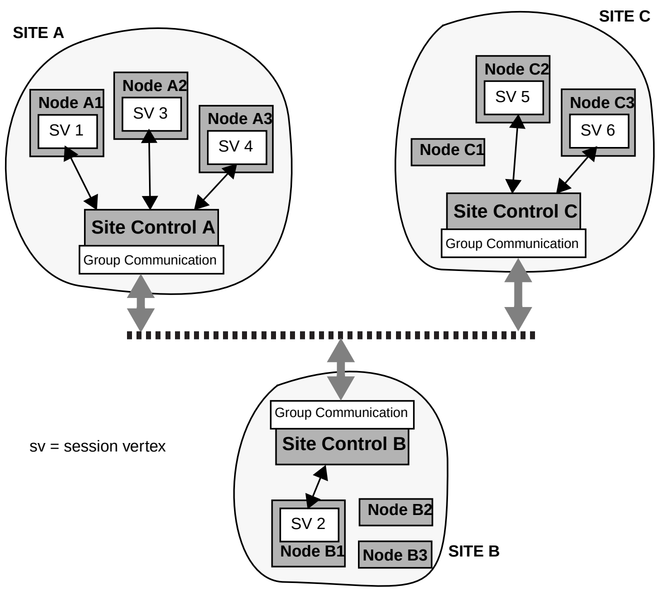


Fig. 3. Application model.

tion management entities. The resulting control architecture is thus semi-distributed: control is centralized on the level of a local network, but distributed on the level of the interconnection network.

III. Platform Architecture

This section describes the platform architecture. It introduces the main abstractions and an application model and discusses the main building blocks of the platform from a functional point of view. The description of implementation details is left to a later section.

A. Sites and Nodes

For the total amount of tightly coupled equipment within the network of a project partner the abstraction of a *site* is introduced. The abstraction of a *node* is introduced as the application dependent mapping of equipment onto a logical application endpoint. Connection and session control within a site is performed by a central entity that knows about the application specific node mapping from a configuration file. Fig. 2 shows a possible node mapping for the personal workplace. The node shown uses different workstations for audio and video processing and for the actual application process. The GUI of the application is displayed on a terminal rather than a workstation screen. Video is displayed separately from the GUI on a second screen. The media input and output devices in Fig. 2 have the logical names *PersMicrophone*, *PersSpeaker*, *PersCamera* and *PersScreen*. Such names are used by the application to denominate connection endpoints. The site configuration file contains for every node a list of endpoint entries, with each entry containing a logical name and its mapping onto a physical address. This configuration information is used by the connection management for the establishment of audio and video connections. Logical device names are in general application specific; they describe the context in which a device like a camera or a microphone is used within a specific application, and they can be as exotic or unique as the application itself. The logical device names shown in Fig. 2 are likely to be employed by more than one application, simply because the node configuration itself is quite common. To illustrate the concept of logical device names, we could add to this node a camera that captures the view from a laboratory window, and call it *WindowCamera*.

B. Application Model

The Beteus application model introduces the abstractions of a *session*, a *session vertex* and a *session application*. A session is the abstraction for one instance of a distributed application that runs on top of the platform. A session comprises, from a logical point of view, a set of nodes as session members. From a computational point of view, a session consists of a set of session endpoints, called session vertices, which are processes that run on the session nodes. The ensemble of session vertices within a session constitutes the session application. In the following, we will use the term session application interchangeably with application or application scenario. If we want to refer to a process running at a node within the framework of a session application, we will explicitly refer to it as session vertex.

Participants are humans or groups of humans that register their name and node with the platform. Once registered they can participate in sessions. For every session in which they participate there will be a session vertex running at their node. Note that it is the session vertex rather than the person that is the actual session participant; the human participant appears as an attribute or name tag of the session vertex.

Fig. 3 shows three sites with each of them having three nodes defined in the site configuration file. An application is indicated that spans all three sites, with three nodes being involved at site A, one at site B, and two at site C. In fact, there is no limitation on the location of the nodes that form a session; they can all be within a single site, or all within different sites. It is therefore also completely hidden to the session vertex on a node if the session in which it participates spans remote sites or if it is local. Session vertices always interact with their local site control, but the processing of a session vertex request may trigger inter-site communication, which is the case whenever connections need to be established in-between sites. The group communication module indicated in Fig. 3 provides the messaging services required for inter-site communication.

C. Site Architecture

The three principal layers of the site architecture are depicted in Fig. 4. The top layer is an application layer containing a generic control panel and application processes - the session vertices. In the middle there is the site control layer which comprises the site manager, the connection manager and the station agents. The site manager implements the functionality offered at the API, whereas the connection manager performs physical connection establishment in collaboration with the station agents. The communication layer finally contains the audio, video and application-sharing software as well as the group communication entity that supports the exchange of control messages between site managers and between connection managers.

The shaded architecture components in Fig. 4, i.e., the site manager, the connection manager and the group communication entity, have only one instantiation within a site. Station agents on the contrary are daemons that are found on every machine on the site network that may be source or sink of audio or video connections or that may run application-sharing software.

D. Site Management

The site manager offers the platform services to the session vertices that run on top of it. Platform services are

• session management

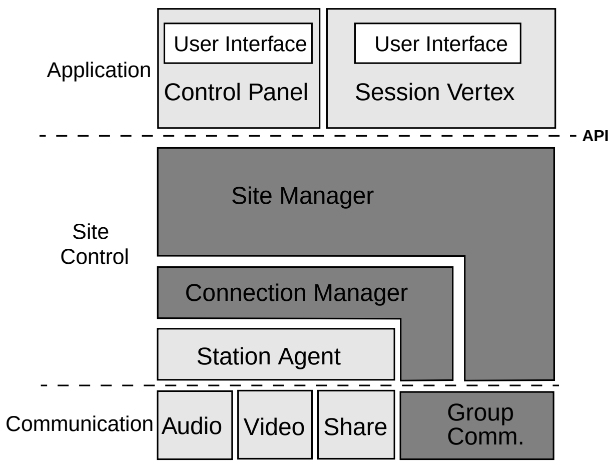


Fig. 4. Site architecture.

• connection management
• endpoint control
• application sharing
• messaging service
• directory service

Some of the services offered by the platform will only be used by the control panel, and others only by the session vertices, although there is theoretically no such limitation.

Participants register with the platform via the control panel. The site management keeps a list of all registered participants and of all ongoing sessions. Participants can create new sessions or join ongoing sessions. When a participant creates or joins a session the control panel forks the session vertex that corresponds to the session application. In case the session is created, the forked session vertex will automatically become the *session master*. The session master has certain rights with respect to the session that other session vertices do not have. This includes for instance the right to delete nodes from a session, or to kill the session. The session master is also the coordination center for the distributed application; it is the session master who configures the session at the beginning and who initiates connection structure changes later on. Other session vertices communicate with the session master via the messaging services of the platform. Most of what the session master does will be in direct response to messages received from other session vertices, or to input from the local GUI. The session master role can be transferred to another session vertex, which is especially necessary when the participant whose session vertex bears this role wants to leave the session. Note that the session master functionality is not necessarily visible at the GUI of the respective session vertex - this is an application design choice.

E. Connection Management

The connection manager receives connection control and endpoint control requests from the site manager. Connection control comprises connect and disconnect requests and is only performed by the connection control of the session master; endpoint control stands for the setting of device parameters like audio volume and video saturation.

The connection manager maps the logical endpoint names in site manager requests to physical addresses. Endpoint control requests can then be forwarded to the corresponding audio or video processes. Connect requests result in immediate connection establishment if all of the connection endpoints are local. If an endpoint is remote, the connection manager of the session master asks the remote connection manager to establish the respective endpoint. The site manager requests only point-to-point connections from the connection manager, but every connect request is accompanied by a hint as to whether the respective connection is part of a multipoint connection structure, in which case the connection manager may use IP multicast [6] if available.

F. Inter-Site Communication

Both the site manager and the connection manager communicate with remote peer entities, as is indicated in Fig. 4.

Site managers need to communicate as part of the directory service, the messaging service and the session management service. The communication between site managers is asynchronous and consists of the reliable transfer of a message from one site manager to one or more other site managers. As part of the directory service, a site continuously broadcasts to all other sites a list of logged participants and a list of sessions of which it is the master site. The messaging service of the site manager allows session vertices to send messages to one or more other session vertices, requiring the reliable delivery of a message from one site manager to a limited set of remote site managers. The session management service deals with session membership. A site indicates a new session vertex with a join message to the site management of the session master. The session master returns a message containing the session membership list to the joining site and an update message to the other sites in the session. The procedure for leaving session vertices is similar.

Connection managers need to communicate in order to establish inter-site connections. As for now, connection endpoints are established sequentially. The connection manager sends an establishment request and receives an acknowledgment once the remote connection manager has established the endpoint.

The communication requirements raised by the site manager and the connection manager are optimally addressed by the Reliable Multicast Protocol (RMP) [7]. RMP supports the reliable delivery of messages to all members of a group with different levels of service ranging from unreliable delivery to totally resilient delivery. It runs efficiently on IP multicast, but allows group members that are not multicast capable to be reached via UDP.

IV. Major Connection Abstractions

This section presents the abstractions that are used at the application programming interface to describe connections and connection structures. Connection types are audio, video and application-sharing.

A. Roles

An application scenario is implemented within a single executable. The session vertices of an application are therefore identical in terms of code, but they behave according to dynamically taken or assigned *roles*. The already introduced master role and a general *participant* role are the only roles which exist by default - all other roles are defined by the application itself. An application may define as many roles as it wishes to, and session vertices may also hold multiple roles at the same time. A session vertex will adapt the GUI that it produces to the role or roles that it takes. Roles fall into two categories: *static* roles and *transient* roles. A static role determines the main behavior of the session vertex and is usually not transferred to another session vertex. Examples for such roles would be the professor role and the student role in the Betel tele-tutoring scenario. Transient roles are created, assigned and deleted as needed; they model whatever ephemeral position a session vertex may have with respect to other members of the session. An example for this would be the role of a momentary speaker in a panel discussion. The application programming interface itself does not differentiate between static and transient roles. This is more a concept that the application designer needs to keep in mind when analyzing an application scenario.

Applications use role names rather than session vertex names or IP addresses to define the endpoints of a connection structure. An application specifies audio, video and application sharing connection structures once on session start-up; later on it will transfer roles inbetween session vertices when it wants to change the connection structure. A typical example for this would be the aforementioned speaker role at the root of an audio and a video multicast connection. The infrastructure will automatically rebuild this multicast connection whenever the speaker role is passed from one session vertex to another.

B. Bridges

The introduction of the role abstraction already provides considerable comfort for application development in that it allows to group connection endpoints. In addition to this the platform provides abstractions for connection structures. A *bridge* is a single-medium connection structure among session vertices. A bridge has source and sink endpoints that are given as role names. The nature of the bridge is determined by the cardinalities of the roles at its endpoints, and may be anything between a point-to-point and a multipoint-to-multipoint connection structure. Up to now it was not necessary to introduce another endpoint addressing scheme than the role-based one. The role-based addressing scheme might become awkward when an application scenario employs an excessive number of point-to-point connections, but no such scenario has been identified until now.

The concept of a medium bridge hides the underlying network from the application. The connection management realizes bridges with whatever transport the network offers. It knows the connection types and is thus able to handle media specific endpoint issues. In a multipoint-to-multipoint audio bridge it will automatically establish an audio mixer at every sink node, whereas it will launch separate receiver processes for every stream in the case of an equivalent video bridge.

The bridge abstraction can also be applied to X11 application sharing. The majority of shared window systems intercept the traffic between an X11 client and server, which allows them to replicate the GUI of the application at various displays by duplicating the client's drawing requests towards the connected servers and by combining events evolving from these servers into one event stream towards the client [8]. A bridge models the group of endpoints on which the GUI of an application is replicated, with the client application as source endpoint and the remote displays as sink endpoints. The actual connection structure it represents is a combination of point-to-multipoint (drawing requests) and multipoint-to-point (events).

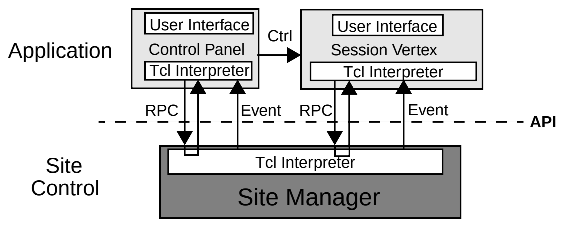


Fig. 5. Procedure calls and event notification at the platform interface.

C. Bridge Sets

A number of bridges, typically an audio and a related video bridge, can be assembled to form a *bridge set*. An application configures the platform on session start-up with a description of the bridge sets that it uses. During the session, only one bridge set can be active at a time. If an application changes the active bridge set, the infrastructure will tear down any connection of the old bridge set that is not included in the new one, and establish the connections that are missing.

The number of bridge sets an application defines corresponds to the number of fundamental application states which in turn corresponds to different temporal phases a session traverses during its lifetime. The programming interface does not directly support the notion of application state, but application state is, like static and transient roles, a concept that the application designer has to be aware of.

V. Application Programming Interface

The API is based on synchronous remote procedure calls (RPC) and asynchronous event notifications as is indicated in Fig. 5. The RPC package chosen for the communication between session vertices and the site manager is Tcl-DP [9], the distributed programming package for Tcl/Tk [10]. Since simple applications will mainly deal with GUI issues, it is possible to implement them completely in Tcl. More complex applications will have C or C++ code in addition to the Tcl/Tk GUI script; they will use the C library of Tcl-DP to call site manager procedures or to register callback functions for event notification.

The API procedure calls [11] are grouped into the following categories:

- **Registration**: user registration and deregistration
- **Endpoint handling**: audio and video device control
- **Session directory**: directory service related calls
- **Session information**: convenience calls
- **Session control**: session membership and lifetime control
- **Bridge set handling**: changing the active bridge set
- **Messaging**: communication among session vertices
- **Role handling**: role assignment and removal
- **Application sharing**: X11 application sharing

The convenience calls allow session vertices to query the session configuration. Session vertices do not maintain records about actual role assignment, actual bridge set or session participants; they retrieve this information from the site manager as they need it.

The main event notifications are:

- **Receive**: a message from another session vertex
- **Join**: there is a new session vertex in the session
- **Left**: a session vertex left the session
- **Kill**: the session got killed or disrupted
- **RoleAdd**: a role is assigned to the session vertex
- **RoleDel**: a role is removed from the session vertex

API procedure call usage and event occurrence are illustrated in Fig. 6. Two session vertices A and B are shown; A creates a session that is joined by B. This session is killed by A when B leaves again. The lifetime of a session stretches from the point of time when it is announced to the point of time when it is killed. The three principal states of the session are *announced*, *initializing* and *ongoing*. An announced session is a session that is scheduled for a certain date and time in the future. Announced sessions are visible via the directory service and help people discover each other's activities. The announcement phase can be skipped by calling SessionInit right after SessionAnnounce. The SessionInit call marks the beginning of the initialization phase where the creator of the session configures the site manager for the actual session application. Initialization comprises role, bridge and bridge set definition. Roles have to be defined before bridges since role identifiers are necessary to specify bridge endpoints. For the same reason, bridges are defined before bridge sets. With the SessionStart call the session enters the state ongoing where it can be joined by other session vertices. This call contains as parameter the initial bridge set identifier. The session creator becomes the first session member and gets automatically the session master role assigned. If he takes additional roles he will assign them to himself with AddRole calls. The session master then has to wait for others to join the session. As is indicated in Fig. 6, B finds out about A's session via a SessionOngoingQuery call. B joins the session with a call to SessionJoin, which is indicated to A with a Join event notification. Connections other than those defined for the general participant role are not established before the session master A assigns a first role to B. The connection structure that is then established between A and B depends on their respective roles and the active bridge set. The example in Fig. 6 continues with a message transfer from B to A that prompts A to change the active bridge set. When B leaves the session, the site management tears down all connections between A and B. The session is formally finished with A's call to SessionKill.

Table 1 shows as example the parameter fields of the DefineBridge API call. The first two parameter fields identify participant and session. The type field marks the bridge as audio, video or shared application bridge. Information granularity is interpreted as window size in the case of video and as sample encoding in the case of audio. Similarly, time granularity is interpreted as frame rate in the case of video and sample rate in the case of audio. Source and sink endpoint names define the logical devices that terminate the connections of the bridge. The call allows further to define a list of role identifiers for sources and one for sinks. The connection type is determined by the car-

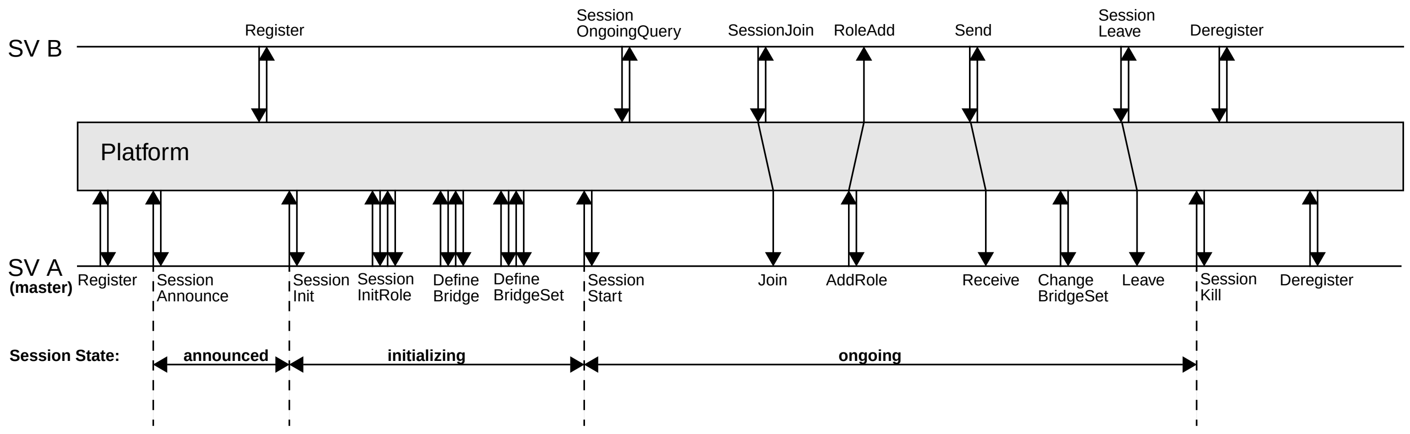


Fig. 6. API procedure calls during a session with two participants.

TABLE 1
THE API BRIDGE DEFINITION CALL

DefineBridge pid sid type ginfo gtime srcepname rslist sinkepname rrlist		
pid	Integer	participant identifier
sid	Integer	session identifier
type	Enum{1,2,3}	1=audio,2=video,3=sharedXapp
ginfo	Integer	information granularity [0..100]
gtime	Integer	time granularity [0..100]
srcepname	String	source endpoint name
rslist	Integer	list of source role identifiers
sinkepname	String	sink endpoint name
rrlist	IntegerList	list of sink role identifiers
returns: bid	Integer	bridge identifier

dinality of source and sink roles within the session:

- **no connection**: no session vertex holds any of the source roles, or no session vertex holds any of the sink roles.
- **point-to-point connection**: one session vertex holds one of the source roles, and one session vertex holds one of the sink roles.
- **point-to-multipoint connection**: one session vertex holds one of the source roles, and multiple session vertices hold one of the sink roles.
- **multipoint-to-multipoint connection**: both source and sink roles are held by multiple session vertices.
- **multipoint-to-point connection**: only one session vertex holds one of the sink roles, and multiple session vertices hold one of the source roles.

The DefineBridge call returns an identifier that can consequently be used to include the bridge in one or more bridge sets.

A. Example Scenario

An example shall serve to illustrate how application scenarios are translated into role, bridge and bridge set definitions. Imagine a distributed school with professors and students all geographically dispersed. Professors have application scenarios for all kinds of teaching purposes at hand, among them a scenario that supports translation work on stage-plays written in a foreign language. The scenario has four states or phases. In a first phase, the professor gives an introduction into the translation assignment that was previously distributed by E-mail. Students see and hear the professor, and they hear each other, which allows them to hear questions asked to the professor by fellow students. In a second phase, the students start to work on the translation of the stage-play. The professor goes from student to student and answers their questions. The editor of the current student is automatically shared with the professor. The professor may return to phase one if one of the questions is of general interest. Once students have finished the translation, phase three begins where students present their results. The professor and the momentarily presenting student are visible to all other students and to each other. The editor of the student is automatically shared with all others, and audio is like in phase one. In phase four, students take roles in the stage-play and recite them. Their image and voice is distributed to the professor and to the other students. The professor finishes the course with some remarks, with the application being again in phase one. During the whole session the professor has as the replacement of a classroom-view an icon-sized video image with low frame rate from every student.

The roles that can be identified in this scenario are:

- **professor**: static professor role
- **student**: static student role
- **studentSpeaker**: visible students in phase two, three, four
- **master**: held by the professor
- **participant**: professor and students

The transient role studentSpeaker is assigned to the visited student in phase two, to the presenting student of phase three, and to the acting students in phase four.

The bridges that need to be defined are shown in Table 2. The first audio bridge is the all-to-all audio of phase one and three. Audio bridge 2 and video bridge 6 form a bidirectional audiovisual connection for phase two. Audio bridge 3 and video bridge 7 form the virtual stage of phase 4. The multipoint-to-point bridge 5 represents the icon-sized classroom view.

TABLE 2
EXAMPLE BRIDGE DEFINITIONS

No.	Medium	Source Roles	Sink Roles
1	audio	participant	participant
2	audio	professor,studentSpeaker	professor,studentSpeaker
3	audio	studentSpeaker	participant
4	video	professor	student
5	video	student	professor
6	video	professor,studentSpeaker	professor,studentSpeaker
7	video	studentSpeaker	participant
8	sharedX	studentSpeaker	professor
9	sharedX	studentSpeaker	participant

Four bridge sets are defined according to the four phases of the application scenario:

- **bridge set one**: audio bridge 1, video bridges 4+5
- **bridge set two**: audio bridge 2, video bridge 5+6, sharedX bridge 8
- **bridge set three**: audio bridge 1, video bridges 5+7, sharedX bridge 9
- **bridge set four**: audio bridge 3, video bridge 5+7

Connection control during the session consists of changing between bridge sets and assigning the transient role studentSpeaker.

B. Application Development

The example scenario already illustrates some aspects of application development on top of the Beteus API. Starting point is the invention of an application scenario. Then comes a problem analysis phase during which the roles, bridges and bridge sets within the application scenario are identified. This is an iterative process because the analysis of the scenario will likely influence the scenario itself. The following design phase comprises the dimensioning of bridge parameters, the specification of the messages that are exchanged between session vertices, and the specification of the functionality to be put into the graphical GUIs. The final implementation phase is mainly concerned with the development of the GUIs. The tight match between scenario analysis methodology and API functionality reduces greatly the effort needed to implement the connection management part of a teleconferencing application.

VI. IMPLEMENTATION

This section describes the implementation of the Beteus platform and of the applications that have been developed so far. Platform and applications run on Sun workstations under SunOs 4.1.3 and are developed in C++ and Tcl/Tk.

A. Audio and Video Transmission

Both audio and video are built on top of the User Datagram Protocol (UDP). One of the biggest issues in Beteus was how to implement multipoint communication with audio and video. It turned out that the structure of the Beteus network is a hostile environment for IP multicast, which would be the natural choice for multipoint communication. This is why the audio and video sender components implement simple stream duplication in addition to IP multicast. The connection managers that control the establishment of connections over the network may employ whichever scheme is possible.

B. Audio

The multipoint nature of audio in Beteus requires some sort of processing on multiple incoming audio streams at the receiving side, which can be either stream selection or mixing. Both stream selection and mixing require silence detection at the sending side and support for talkspurt transmission at sender and receiver. The Beteus audio component implements silence detection at the sending side with an adjustable threshold value and is built on top of the Real-time Transport Protocol (RTP) [12], which in turn uses UDP for transmission. The receiving side supports both stream selection and mixing. Both sender and receiver generate activity events that can be graphically displayed on the GUI. The sender indicates begin and end of talkspurt to the local user, whereas the receiver indicates activity for each of the incoming streams on which it listens. The two audio encodings that are supported are 8kHz/8bit and 16kHz/16bit.

C. Video

Video transmission is built around the XVideo board from Parallax. The compression of the Parallax board follows the JPEG standard for the compression of still images [13]. On connection setup, the video sender allows to specify a target data rate that is consequently enforced by means of a control loop in which maximum and measured data rate are constantly compared, with the JPEG compression factor being modified according to the result of this comparison. Such a mechanism was clearly necessary in the case of Beteus where there are data rate restrictions per video stream and traffic policing within the network. The video receiver adapts automatically to the actual compression factor as it also adapts to frame rate and window size. Care was taken to have a constant frame rate within the receiver window because the human eye is extremely sensitive to frame rate irregularities. The benefit of this is that the subjective quality of the Beteus video component is excellent even at frame rates as low as five frames/s.

D. Application Sharing

The application sharing component of the platform is Xwedge from project partner ETH Zürich [8]. Xwedge is a distributed shared window system that has agents running at all implicated client and server sites. X11 clients connect to the local Xwedge agent which in turn communicates via the Transmission Control Protocol (TCP) with remote agents and the local X11 server. The Beteus API offers three calls for application sharing control: a session vertex can get a list of sharable applications, which are the clients that are momentarily connected to the Xwedge agent, and it can share and unshare an

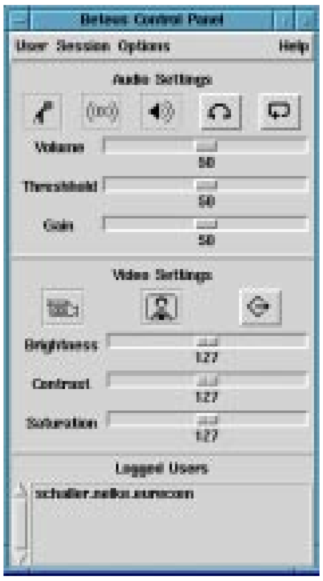
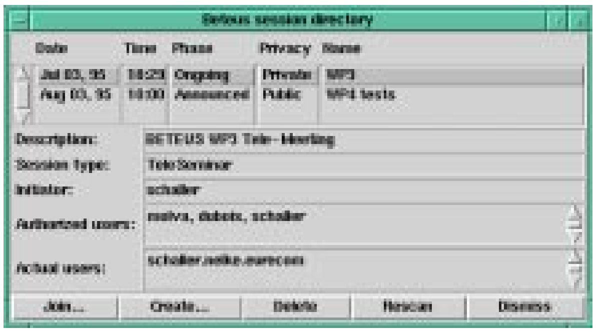


Fig. 7. Control panel user interface.



Date	Time	Phase	Privacy	Name
Jul 03, 95	18:29	Ongoing	Private	MP3
Aug 03, 95	10:00	Announced	Public	MP4 tests

Description: BETEUS MP3 Tele-Meeting
Session type: TeleSeminar
Initiator: schaller
Authorized users: molva, dubois, schaller
Actual users: schaller.nelke.eurecom

Fig. 8. Directory service window of the control panel user interface.

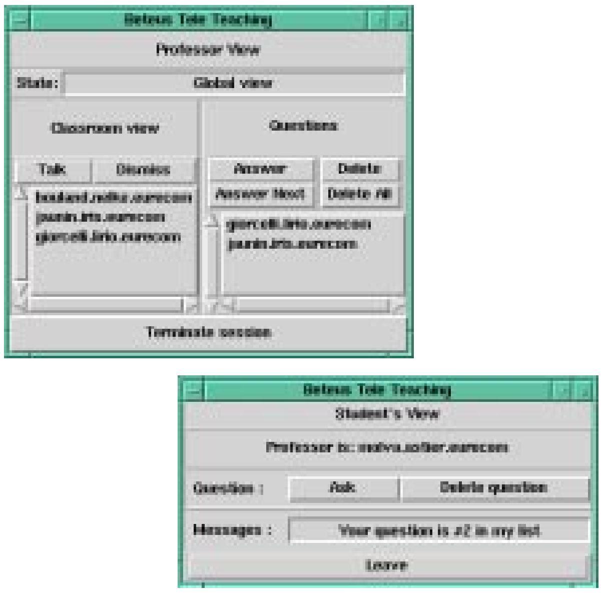


Fig. 9. The user interfaces for the tele-tutoring application (professor/student).

application. Sharing means that the interface of the chosen application is replicated at the sink endpoints of the currently active X11 bridge. As for now, the platform does not implement the rich set of floor control mechanisms that Xwedge offers. The platform uses the default floor control mode where the floor follows mouse clicks and keyboard input.

E. Site Control

The persistent components of the platform are the site manager, the connection manager, the group communication component and the station agents. A station agent launches audio and video processes and Xwedge agents and relays operation requests from the connection manager to these processes and operation results and asynchronous events back to the connection manager. The connection manager is forked by the site manager and establishes TCP control connections to all station agents when coming up. The site manager exists in two versions: the normal runtime version and a development version. The development version of the site manager forks a dummy connection manager that reads the site configuration file and returns positive responses to site management connect and disconnect requests. This allows to test session vertices in emulated sessions on a single screen and without establishing audio and video connections.

F. Applications

The applications that have been developed so far are the generic control panel, a tele-meeting application and a tele-tutoring application.

Fig. 7 shows the main window of the control panel with the audio and video device control fields and the list of logged users at the bottom. From this window the directory service window shown in Fig. 8 can be invoked. The directory service window lists ongoing and announced sessions and allows, among other things, to join or create sessions.

The tele-meeting scenario is a simple framework for work meetings that can be used within many environments. The audio and video connection structure is all-to-all, i.e., everybody sees and hears everybody else. There are simple GUIs for a chairman and a normal participant, with the chairman being able to assign the role of a presenter to one of the session participants. The presenting person can share one of its X11 applications with the other participants. The chairman interface allows to transfer the chairman role to another participant, in which case this participant gets his interface exchanged for a chairman interface. The tele-meeting scenario is implemented with a single bridge set containing an all-to-all audio bridge, an all-to-all video bridge and a one-to-all application-sharing bridge with the presenter role as source. Connection management gets active only when the presenter role is assigned, or when participants join or leave the session, in which case their connection endpoints are added or removed automatically from the audio and video bridges.

The tele-tutoring application is a remake of Betel on the Beteus platform with the difference, that students are geographically dispersed. Fig. 9 shows the GUIs of professor and student. The application can be in the states *global* and *talk*. In the state global, the professor has a video window for every student, and can himself be seen and heard by all students. In the talk state, the professor talks to a single student, but audio and video of both professor and student are distributed to all other students so that everybody can follow their discussion. The student can also share an X11 application to show his work. The roles, bridges and bridge sets defined for this scenario resemble the ones described in the example scenario of Section V. The teletutoring application is still a simple application, but it already has much more connection structure dynamics than the telemeeting scenario.

VII. The Beteus Network

A global view of the Beteus network was already shown in Fig. 1. Fig. 10 provides additional information on the local site configuration used during the tests. Every project partner has an ATM LAN on the basis of a Fore Systems ASX-200 switch [14] that connects via a 34Mbit/s E3 interface to the ATM pilot network and via 100Mbit/s TAXI interfaces to two Sun Sparc10 workstations that together form one node (see Fig. 2).

The only service provided so far by the ATM pilot is the interconnection of bidirectional permanent virtual paths (PVP) or semi-permanent virtual paths (SPVP). The PVP service offers connectivity over a longer period of time, whereas SPVP requires occasional or periodic reservation. Beteus was most of the time using the SPVP service and had connectivity on Tuesday mornings from 8:00 to 12:00.

The issue that came up after the interconnection problem was solved was the topology of the Beteus SPVP overlay network. Two extreme proposals were full interconnection and serial alignment. It was possible to arrange the project partner LANs in a chain starting at Eurécom and going over CERN, EPFL and ETHZ to the TUB in Berlin. While such a configuration would have had a minimal number of PVPs on the ATM pilot and therefore a cost advantage as compared to other solutions, it would have forced every project partner to participate in every test and it would have made a possible point of failure out of every intermediate ATM switch on the line, which is why it was declined. The topology that was finally adopted was the fully meshed network where a given project partner has PVP connections to every other one.

Another issue was the dimensioning of the PVP connections. The calculation of the maximum bandwidth for a PVP is based on the following assumptions about the traffic originating from a Beteus site:

- one low quality video stream (12fps): < 1 Mbit/s
- one high quality video stream (25fps): < 2 Mbit/s
- one audio stream: 64 kbit/s or 256 kbit/s
- control and application sharing: negligible

It was decided to reserve a maximum bandwidth of 3 Mbit/s for every PVP.

A. Multicast

The most important problem to be solved was multicast. It was clear that multicasting had to be done within the local networks of the project partners given that the ATM pilot does not

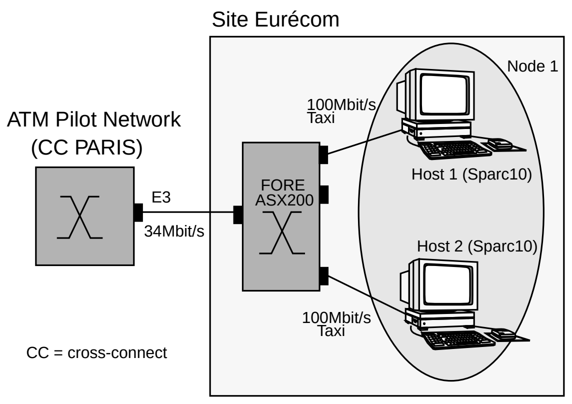


Fig. 10. Site configuration at Eurécom.

support any form of multicasting. At the beginning there was the hope that Fore Systems' proprietary SPANS signalling could be tunnelled through the ATM pilot, and that then either Fore Systems' native ATM multicast or its implementation of IP multicast could be used. This did not work because the then actual ForeThought 3.0 switch software did not support more than one signalling channel at a given network interface port. Since a Beteus ATM switch accessed the ATM pilot via a single E3 port it was not possible to establish signalling tunnels with more than one remote site, which is, there was no way to establish a fully meshed network of signalling channels between the Beteus switches.

ATM-based multicast itself turned out to be hard if not impossible to achieve. The ASX-200 switch does not allow an incoming cell stream to be duplicated into two streams that leave through the same output port, as was required by Beteus. This is a natural restriction if one considers that output ports of switches are usually connected to input ports of other switches or to host interfaces. In both cases there is no need to send one cell stream twice over the same link. A solution proposed by Fore Systems was to let the switch multicast an incoming stream to as many switch output ports as there are final destinations, and loop the streams back into the switch and then finally out via the E3 interface. This proposal was not further pursued because such a solution would have monopolized the Beteus switches of which some where used in parallel for other projects.

The only solution that remained after the ATM-based multicast alternative was discarded was multicast on application level. One possibility for this was to have one central multicast daemon running on one site, with each of the other sites having a point-to-point connection to the daemon machine. Such a configuration is similar to the multipoint control units found in N-ISDN conferencing systems. Another possibility was to distribute the multicast functionality onto all sites and have one multicast daemon per site. None of the two solutions was adopted because it was thought to be awkward to have delay sensitive media streams pass four times through the UNIX user space. The solution that was finally implemented was source-based stream duplication whereby audio and video senders transmit streams to more than one destination. While this multicast scheme missed any elegance, it seemed to be the best of all possible compromises. It might appear to be a drawback of source-based stream duplication that it sends identical streams over the same network link, but note that this would have been equally the case in an ATM-based solution. A real drawback of source-based stream duplication is the extra load it puts on the source machine, but this did not really matter in the case of Beteus where media streams were sent to a very limited number of destinations. The platform architecture takes such performance issues into account and allows machines that are dedicated to audio and video transmission. Stream duplication avoids additional processes and thus additional points of failure, and in terms of end-to-end delay it is almost as optimal as ATM-based multicast.

As a consequence of the chosen multicast scheme and the lack of signalling, Beteus hosts needed to be interconnected via a fully meshed network of permanent virtual circuits (PVC) of which the endpoints were mapped to IP addresses. This made the ATM network pilot transparent to the Beteus platform that runs on top of IP.

B. *Platform Deployment*

A never ending source of problems was the administration of the Beteus overlay network. Every project partner had to manually configure four PVPs, sixteen permanent virtual channels and sixteen IP/ATM address mappings for a trial. At many sites, Beteus interfered with other projects that used the same equipment and that had their own configuration. It often happened that machines were not available for the tests, or were replaced at short notice by others, for instance because of hardware failures. The general configuration file that was used by the project partners was therefore often not up-to-date, or if it was it could happen that switches or interfaces were badly configured. Another error-prone procedure was the reservation of the ATM pilot connections that had to be done for every irregular event. During the project we gained some experience in detecting and solving connectivity problems, and made extensive use of network management tools to this purpose.

The performance of the network was excellent. The round trip time measured between Eurécom in France and the Swiss sites was around 30ms. The Swiss sites among themselves measured round trip times of as low as 4ms [15]. A network management platform tailored to Beteus was monitoring performance at levels from the ATM layer up to the video and audio application processes. An example measurement for the bit loss rate at ATM level is 3.03×10^{-9} [16], which indicates a very reliable network. The quality of the network was directly visible at the application level. Audio and video had low losses and low latency.

The complete platform ran more often on the local test-bed at Eurécom than on the ATM pilot. This was mostly due to the aforementioned time-consuming connectivity problems. As a result of this the main attention of the project shifted away from the applications to basic audiovisual communication. Although the platform was up and running, there was no time to really evaluate and improve the tele-meeting and tele-tutoring application scenarios, as it was foreseen at the beginning. The two scenarios were demonstrated to the European Union as they were originally conceived, i.e., without having traversed a cycle of gradual improvements as it is supported by the platform API. It is therefore also the audiovisual transmission that received most of the feedback during the network trials, and not the platform itself.

VIII. Related Work

The Beteus platform features an API for multipoint teleconferences. Such an API was first found in the Touring Machine developed at Bellcore [17]. The Touring Machine API reflects the limitations of the underlying analog network, and does not offer high-level abstractions comparable to those of Beteus. This is equally true for the API of the LAKES platform developed at IBM [18]. The Beteus, Touring Machine and LAKES APIs are complete in that they offer both session and connection management functionality. More recently, object-oriented frameworks for multimedia middleware have been described that concentrate on connection and configuration management and that are much more flexible than monolithic APIs. One example for this is the Medusa platform developed at Olivetti [19]. Another example is the multimedia system services (MSS) architecture of the Interactive Media Association (IMA) [20]. A platform similar in spirit to IMA, but with a complete session framework, is currently being developed at Eurécom [21]. High-level APIs like the one of Beteus can be implemented as toolkits on top of a lower-level multimedia middleware.

Another important point about the Beteus platform is that it is designed to be deployed on a high-speed WAN with multipoint communication support. It may be compared with standalone applications like ISABEL [22] or JVTOS [23] that were also deployed on the European ATM pilot. What Beteus distinguishes from these applications is that it allows more complex connection structures, and that it allows dynamic changes to the connection structure. Most teleconferencing applications, including those found on the Internet MBone [24], establish static connection structures.

IX. Conclusion

The platform, as it is implemented now, supports conference-style communication among a small number of sites with a static relationship to each other. Examples for such groups of sites are

- universities with a common tele-teaching program
- laboratories working on a common project
- administrative units of an international enterprise.

The platform is not designed for ad-hoc communication on a network with a large number of sites like the MBone. Such a deployment is imaginable, but it would require a redesign of at least the directory service and the group communication component.

The main emphasis of the design of the Beteus platform is on the application. The platform exports an API that considerably reduces the effort it takes to implement a multipoint tele-conferencing application. The platform is also a runtime environment for such applications; it allows applications to run in parallel and offers a directory service that informs logged users about what is happening on the platform.

The main emphasis of the field trials should also have been on the applications, but this goal was not achieved. The experience of Beteus is that the broadband network needs to be much more transparent than it is now in order to support advanced multipoint applications. The configuration effort necessary to allow for connectivity on IP level in a network as small as the one of Beteus was perceived as a major obstacle.

The Beteus platform is now being deployed on France Telecom's ATM WAN in Sophia-Antipolis that became operational at the beginning of 1996. Work on it will continue in various directions. In a general move to shorten communication paths and to increase distribution within a site, we started to reimplement our platform on top of the Common Object Request Broker Architecture (CORBA) [25]. A future version of the API will be defined in CORBA's interface definition language IDL.

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