A Framework of Software-Based PBX Services over Internet[†]

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Abstract

Although Internet is a powerful and economical communication vehicle, its poor quality-of-service (QoS) is still the major obstacle for multimedia transport over Internet. Basically, both the high-speed broadband network platform and the effective multimedia application platform are needed to gain and promote the benefit of multimedia services to end-users. The ATM-based broadband network would be an adequate inter-network platform for multimedia communication. In general, different application platforms will be needed for different traffic requirements. In multimedia applications, a hybrid service is essential for supporting real-time traffic and non-real-time traffic simultaneously, for example in a conference call, data and voice have to be transferred at the same time. The goal of this paper is to design a framework which can support PBX services with multicasting capability for TCP/IP running over Internet and provide portability to the upper-layer distributed applications. A model of the hierarchical service architecture is proposed to solve the problem of QoS for end-users. Finally, we build an agent-oriented framework for distributed switching Agent with object caching mechanism to provide the capability of having a better deployment to clients.

Keywords: *Quality-of-Service (QoS), Addressing Scheme, Multicast Mechanism, Basic Call State Model (BCSM), Call Control Object, Session Management, Distributed Switching Agent.*

1. Introduction

Both conventional telephone system and ISDN system have claimed their supports of multimedia services to the world. The multimedia-integrated services, such as the hyper-text retrieval system, also have offered much contribution to the office environment. However, there remains the challenge issue of advanced multimedia transport technology. The perceived solution is to have a single integrated network transports all kinds of information. Following the wide use of Internet and the success in high-speed broadband communication networks, multimedia services are becoming promoted to all users, however, users of packet-switched networks such as Internet may experience longer delay without guaranteed QoS. A great deal of research effort has been emphasized on these problems by enhancing either the communication protocols, e.g. RSVP[1], or the switch architectures, e.g. the IP switch. The other group strongly believes that the ATM networks will fully offer far more bandwidth and capacities than most users' need. QoS control is important in the Internet to be capable of supporting real-time voice services. Concerning the QoS guarantee for a software-based PBX system, the difficulty lies in the fact that the end-to-end delay is experienced for 300 ~ 500 ms in the Internet. Supporting QoS is simple and easy in circuit-switched networks with fixed bandwidth management, however, it is quite difficult and complicated in packet-switched networks, such as Internet, requiring dynamic bandwidth management.

In the multimedia communication environment, the design of an addressing scheme for user identification is needed. Besides the conventional addressing rule to be followed, a global addressing scheme is necessary in the distributed networking environment. Thus, applying the addressing scheme of domain name server (DNS) to inter-networking group of user's assignment may be a good choice. We will also use the address resolution protocol (ARP) [2] as the address mapping scheme of endpoint identity. Before the end users start conversation with each other, connections must first be established following the

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call procedure defined in the standards of TCP. Communication with multicasting capability may be achieved easily in connectionless services such as IP protocol with multicast group address, i.e. class D [3] IP addressing, but it is complicated in ATM connection-oriented services without native multicast supports. For the inter-networking environment, the multicast mechanism must have address scalability, be capable of reducing transmission delay and the aggregation of multiple connections.

The object-oriented software component technology[4] is becoming mature, meanwhile, new examples of Java applications have also been rapidly developed. Therefore, our discussion will cover the connection aspects of networking as well as the technical issues involved in the upper-layer design. The open distributed agent-oriented software components, such as multicast server and endpoint, are implemented in the LANs/WANs to exhibit that the agent-oriented software component technique will be a practical implementation in the Internet. Our investigation is in the context of a novel hierarchical system model which is capable of sensing all users with global addressing within the system. The multimedia services are considered as composed of two levels: the physical level performs the physical functions of raw stream transferring and multimedia stream rendering; the service level provides the service of call control and session management.

The remainder of this paper is organized as follows. In section 2, we present the system architecture for multimedia services over Internet and two addressing schemes for the support of registry of end-users. In section 3, we explore the buffer management and bandwidth management for multipoint conferences. Section 4 discusses the aggregated session mechanism to be used in the multicasting connection management, and section 5 concludes this paper.

2. System Architecture

The system architecture of the framework of software-based PBX services consists of three layers as shown in Figure 1.

Layer 1: The addressing server (AS) performs the address mapping for each endpoint, the registry of multicast group membership and the function of transferring the logged-in users to the multicast server (MS) for establishing the connections to each endpoint user in the system.

Layer 2: The MS offers call forwarding and manages all sessions of conference calls of more than one user.

Layer 3: The endpoint (EP) provides the function of basic call connection and multimedia interaction. Each endpoint can access to both the AS and MS as a client, and a gateway server (GS) can also be configured at this layer to provide the functions of signaling and interface to the external legacy devices such as facsimile machine, media player or phone set.

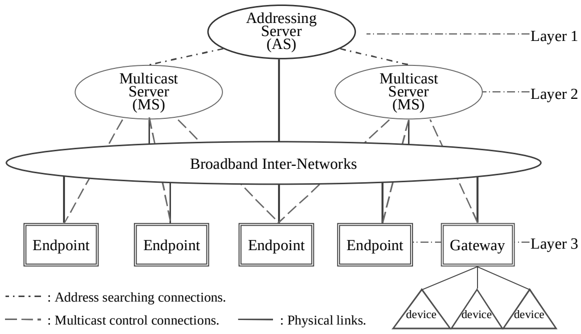

Figure 1. System architecture

All entities reside on the system are built as agent-based software components to provide the capability of having a better software distribution path over Internet.

2.1. The Addressing Scheme

The AS has to provide global addressing capability so that each endpoint can access its own AS from any visit system which can access to the globally specified AS via Internet. Both searching method and accessing scheme of the endpoint identity are described as follows.

(1) Searching Method

In the distributed dynamic hashing scheme proposed by Jain et al. [5], the homogeneous hashing rule must be followed to search and allocate the ASes (or called translation servers) when the endpoints grow up to the limitation of system. The key consideration here is the requirement of accessing in heterogeneous distributed system. The hashing rule is difficult to be followed by each request of endpoint searching. The AS must provide an active searching method to forward the searching request to others when the endpoint entity does not exist in the AS. There are two relevant forwarding methods:

Ring search method: A logical ring [8] is formed by traveling an Euler path on the network of ASes. When a node accepts an incoming request from a remote endpoint, it will start searching the internal database. If no answer is received, the request message is forwarded to a neighbor node in the specified path.

Spanning tree search method: N-1 connections must be established simultaneously for a spanning tree [7] while the searching function is executed to access the home address of each endpoint user.

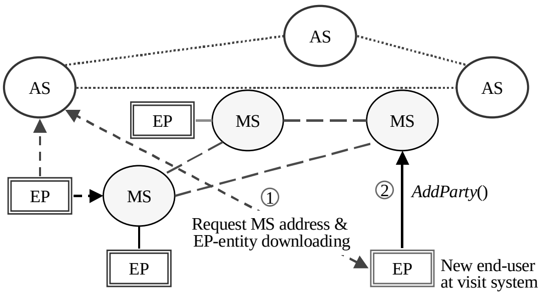

Figure 2. A visit accessing scenario

The selection of the above searching methods is based on the network infrastructure to achieve good performance. The searching effort measured by all addressing paths is dependent on the structure of the node connection related to the registration of endpoint users. Basically, there are two constraints: response time constraint and connection resources constraint. Although the spanning tree search method requires more connection resources, it provides less delay and search simplicity in our framework.

(2) Accessing scheme

The AS has to provide global addressing capability so that each endpoint can access its own AS from a visit system which can access to the globally specified AS via Internet. Therefore, the AS needs to support the following two accessing scenarios:

Home Access: The AS provides the directory service, any endpoint can access the AS via the inter-network with its logical name (or symbolic name).

Visit Access: When a user visits at any visit system, he/she can use that system as an endpoint to register to his/her home AS. A user profile and an MS address near the visit location will be sent from the home AS to the EP in the visit system. Figure 2 shows an example of visiting at a visit system. The user first makes a home access from the visit system to request for the address of an MS and download the endpoint-entity from the AS, and next requests for a connection to the nearest MS assigned by the AS to start the multimedia service.

2.2. Address database management

Each AS supports a database of endpoint id. It is difficult to find a good solution for a large database accessing within a limited access time interval. In general, B-tree and hash accessing schemes are two trivial methods [6] used in the mass data accessing environment. The decision for the number of indexing pointers in each B-tree node is a key issue that will directly affect the performance of endpoint id searching in the whole system. The problem of the overflow control area for hash accessing is dependent on the number of ASes and the number of endpoint id entries in each AS.

2.3. Multicast Scheme

The MS has the capability to multicast call connections to all endpoints in a conference call. It also controls the endpoints that are actively multicasting packets and all of the endpoints in the conference transmit their packets to the MS. Many multicast routing algorithms based on a spanning tree scheme have been proposed [7][8]. In order to shorten the transmission delay, we propose a full mesh multicast scheme to handle *n* active MSes. Each MS maintains *n*-1 connections to connect to its neighbor MSes in the conference, and these connections can be shared to reduce the connection resources while the MS serves multiple conferences at the same time. When a new MS wants to join an existing conference call, it has to obtain the MS address list of the conference from one of MSes which is already in the conference. The new MS will use the address list to establish individual point-to-point link to other MSes. A hop count field must be included in each packet and initialized by each endpoint with 2, the hop count will be decreased by 1 when a packet is routed. Finally, the packet with hop count 0 is sent to the endpoint and is then discarded by each MS. The strategy that a hop count 1 is always forwarded to the next MS but dropped when the count is zero is used to avoid the loop problem of packet forwarding. Traffic class and sequence number fields are also included in each packet for in-sequence transmission, both fields are recognized by each receiver in the hop-by-hop transmission. When the service is of non-real-time traffic, the sequence number will be used to perform the ordered transmission with reliable error control protocol. For real-time traffic, error control is not required, but the sequence number is required to distinguish from earlier packets to be dropped.

3. Traffic Control

The MS provides a conference call among two or more endpoints and the bridging function between two or more multicast group connections. The endpoints in a multipoint conference are not aware of the number of MSes cooperated. In order to satisfy the guaranteed QoS transmission in the application of multipoint conference, strategies of buffer and bandwidth allocation must also be included [8][9].

3.1. Buffer Management

In the multicast connections, the same contents are sent in each session to every receiver. If each session allocates its own buffer to keep their transferred packet, the buffer resource will soon be exhausted. Meanwhile, it not only causes packet duplication but also limits the number of multicast users in the system. Obviously, shared buffering is more efficient in supporting the one-to-many buffer

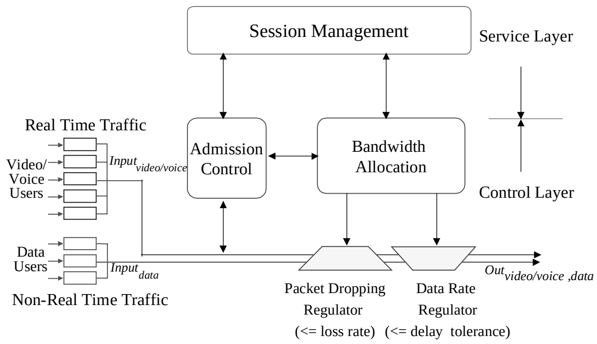

Figure 3. The traffic control mechanism

management [10].

If b is the buffer size in unit for memory transferring, T_{mem} is the rate of memory I/O movement and T_{play} is the rate of voice I/O playing. The total number of multicast users N in the system can be expressed as $N < T_{mem} / T_{play}$. Thus, the total buffer size required for multicast group users is measured by $S = mb$, where m is the number of users in one multicast group and $m < N$. To achieve minimum S, the cost of buffer allocation must be reduced for each multicast group, so that $S < mb$. The two-state Markov chain is used for bursty traffic modeling. The probability that a source is active is $p = \alpha/(\alpha+\beta)$ where the average active (talkspurt) time is about $1/\beta = 0.4\sim1.2$ sec, the average off (silence) time is about $1/\alpha = 0.6\sim1.8$ sec [11]. Therefore, the minimum buffer size S would be considered as $S = pmb$ with packet loss rate of 1%

3.2. Bandwidth Allocation

The performance of data service will be affected when the video/voice traffic is heavy. In order to simplify the analysis of bandwidth allocation, the traffic type is classified as real-time traffic and non-real-time traffic. Video/voice users contribute to the real-time traffic and data users contribute to the non-real-time traffic. The traffic control mechanism used in this paper is shown in Figure 3, and the control strategies must include the following three parts:

Data user management: The bandwidth of data-service users are sacrificed when the voice traffic load is heavy during talkspurt, and can be compensated when the voice traffic is light.

Video/voice user management: Bandwidth reservation can be considered as two phases: the connection phase before starting voice service; and the transmission phase when the voice users are in conversation. In the connection phase, the admission control unit allocates the bandwidth and continues to reserve during the transmission phase. When the bandwidth runs short for a video/voice connection, the bandwidth of a data connection is first considered to be regulated depending on the QoS of the data user. If the spare bandwidth is still not enough for the new video/voice request, the video/voice packets are dropped.

Bandwidth allocation: The function of bandwidth allocation not only provides the admission control but also executes the function of rate regulation. When the admission control unit first receives a request from a user, it checks to see if the bandwidth resource is sufficient. The data rate regulator slows down the data rate when the bandwidth is not enough in bursty state. In contrast, the data rate will be compensated while the real-time users are in silent state. If the bandwidth can not be regulated among data users, the packet dropping regulator will be activated to drop video/voice packets with the specified loss rate.

In the traffic control queueing model, the bandwidths allocated to video/voice and data users are denoted by B_v (bits/sec) and B_d (bits/sec) respectively, and the total bandwidth in the system is denoted by $B_t = B_v + B_d$. $Q_{t\text{-}1}$ and Q_t are the number of packets in queue at measurement cycle t-1 and t respectively. When bursty traffic occurs, the bandwidth B_{Burst} requested by a talkspurt can be expressed as follows:

$$B_{Burst} = B_d + \sum_{i=1, i\neq j}^{N_v} B_{vi} + B_{vj},$$

$$B_v = \sum_{i=1}^{N_v} B_{vi}, \; N_v \leq N, \text{ and}$$

$$B_d = \sum_{i=1}^{N_d} B_{di}, \; N_d \leq N.$$

Where B_{vj} is the bandwidth requested by voice user j in talkspurt state, N is the total number of users, $N = N_v + N_d$, N_v and N_d are the number of real-time users and data users in the system respectively. Let B_{mean} be the mean rate of voice packet transmission and B_{peak} be the peak rate of packet transmission. A fraction of $(1 - B_{mean} / B_{peak})$ is the maximum bandwidth allowed to degrade the data rate and to increase the current bursty voice channel. $B_{drop\text{-}off}$ is the maximum amount allowed to drop the voice packet. The dropped term includes the part of $(Q_{t\text{-}1} - Q_h)$ packets which exceeds the threshold Q_h during the measurement time interval Δt. B_{drift} is the maximum drift of delay for data transmission and B_{loss} is the maximum bandwidth when the voice packet dropping rate is 1%. The measured values of data rate regulation and packet dropping for bursty traffic are given by

$$B_{regulated} = (1 - B_{mean} / B_{peak}) \times B_{burst},$$
$$B_{drift} = B_{drift(t\text{-}1)} + B_{regulated}$$
$$B_{drop\text{-}off} = \min(\max(0, B_{Burst} - B_{mean} - B_{regulated} + (Q_{t\text{-}1} - Q_h) / \Delta t)), B_{loss}),$$

When $B_{regulated} > 0$, the data rate regulator is activated and B_{drift} is decreased by one regulating unit. If $B_{drop\text{-}off} > 0$ and the dropping rate is less than the loss rate in the specified QoS, the packet-dropping regulator is activated to drop video/voice packets. B_{drift} will be increased when

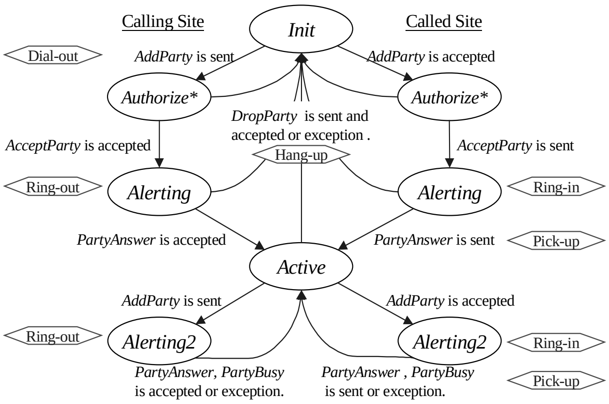

Figure 4. The modified call state model

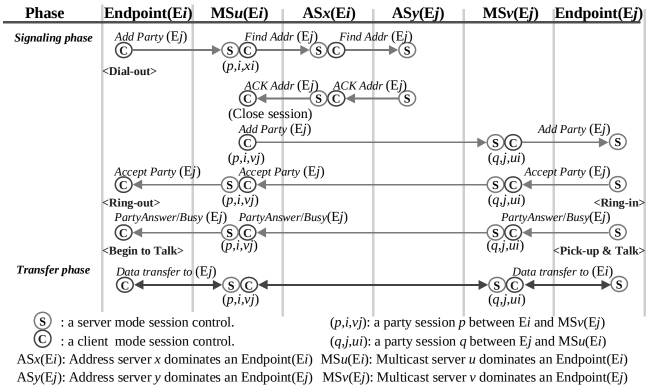

Figure 5. Session control protocol

the data rate is compensated.

4. Session Management

A session management manages the interactions among all connections while these calls are in-progress. According to the call state model, the session management sets up call connections, distributes call objects using session control protocol and multiplexes messages into an aggregated call connection.

4.1. The Call State Model

The basic call state model (BCSM) in the ITU-T recommendation is a high-level finite state machine [12][13]. It is useful in the description of call control functions for managing the basic call handling and call forwarding in PSTN and intelligent networks (IN). The BCSM includes both the originating mode and the terminating mode. In the originating mode, before the caller and the callee start talking with each other, the caller must complete the state to check through the *Collect-Information* state, *Analyze-Information* state, *Route-and-Alerting* state and *Active* state. The callee also has to go through all the states in terminating the call state model successfully. In our system framework, the *Collect-Information* and *Analyze-Information* states are merged into a single state due to that the endpoint has to provide more intelligent processing capabilities to simplify the call procedure to users.

The modified call state model is shown in Figure 4, where the initial state *Init* is ready to accept the *Dial-out* event from every call process. This state will change to *Authorize* state when either the *AddParty* command is sent by the caller or an incoming *Addparty* command is accepted by the callee. The state transition continues to the *Alerting* state if the *AcceptParty* command is either sent by the callee or accepted by the caller. The callee has to respond to the *Addparty* command, then the *AcceptParty* command is sent back to the caller and the *Ring-in* event will occur at the called site. After the *AcceptParty* command is accepted at the calling site, the *Ring-out* event takes place at the called site. The events, *Ring-in* and *Ring-out*, persist in the *Alerting* state till either the *Pick-up* event is respond by the callee or an exception event occurs. The *Active* state means that the first call request is completed and another call request can start for a conference call. The *Alerting*2 state is used to distinguish from the *Alerting* state while the new callee is inviting in the conference call. More details of the call control and interactive protocols are described in the following sections.

4.2. The Distributed Control Call Objects

The call is first issued by the endpoint or the gateway server and switched by the MS to the destined endpoint to which the connections are established hop by hop to. The distributed control call objects are also created in both ends of the connection to handle the signaling of the calling state until the call is completed. In a conference call, the call objects have to be distributed over several MSes, but some redundant call objects will be merged to reduce the multicast overhead from different TCP connections.

4.3. Session Control Protocol

In the case that a connection is shared among multipoint communication[14][15], it must be maintained for the endpoint users during signaling and transferring phases. The session control protocol is depicted in Figure 5. When the signaling phase is completed, the transferring phase is continued with the same connection. Therefore, the overhead for connection establishment is saved and the failure probability of new connections for the transferring phase is ignored. In the software design of AS, MS, endpoint and telephony server, there must be a kernel to handle the client-server connection, and is implemented by agent-oriented software components. All the components will inherently commit the client-server connectivity to agent-oriented communication in broadband networks.

4.4. Session Association

The relation among the call connections, both the call object and call party group are the key of accessing to multiple sessions while the packets are switched in the MS. The relation is referred as session association and represented by a vector <Session ID, Group ID, Object Pointer>. Each call connection is composed of several individual sessions, these sessions are organized within a data structure to form the vectors of session association. Thus, the handler of data structure of session association vectors has to be designed for providing an efficient accessing scheme that both data and object are pointer to the complicated structure of multiple sessions. Each single element of session association vectors will be individually treated as an access pointer of its own call instance to handle the multicast services.

4.5. Aggregation of Call Connections

Multiple connections between MSes are established for a number of call object pairs (e.g. twice as many as the connections) to form the call party or conference calls, and the connection resources grow with the call objects distribution between MSes. Thus, the scheme of aggregating connections must be employed to aggregate the traffic of multiple call objects into one connection, and to reduce the redundant requests of call connection.

5. Conclusions

We have designed and implemented the framework of software-based PBX for telephony services over Internet by using the agent-oriented software component technique. We believe that the architecture of agent-oriented software component is a trend to lead the distributed multimedia services in the future [15]. The agent-oriented software component technique is based on the concept of distributed cooperative communication networking over wide-area distributed networks. It is truly an important property for supporting multimedia communication in the Internet. There has the benefit of software distributed processing over Internet because the agent-oriented software component technique is popularly used in the distributed processing environments. Therefore, we also use this technique to implement those components executed at each process unit in an example system of voice telephony. Although there has less example showing multimedia services integrated in broadband networks, the need of integrated service in broadband networks grows rapidly. This trend represents primarily large businesses and many operations for large intra-company multimedia networks. In this paper, the framework of software-based PBX system is emphasized, and the many-to-many multicast mechanism is implemented by the agent-oriented software components to distribute the call objects via the Internet. Further investigation will be conducted to show if it is capable of achieving the efficiency of system operation.

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