Supplementary Services in the H.323 IP Multimedia Telephony Network

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Index words: H.323, H.450, packet telephony, IP telephony, multimedia conference, video conference, audio conference, supplementary services

ABSTRACT*

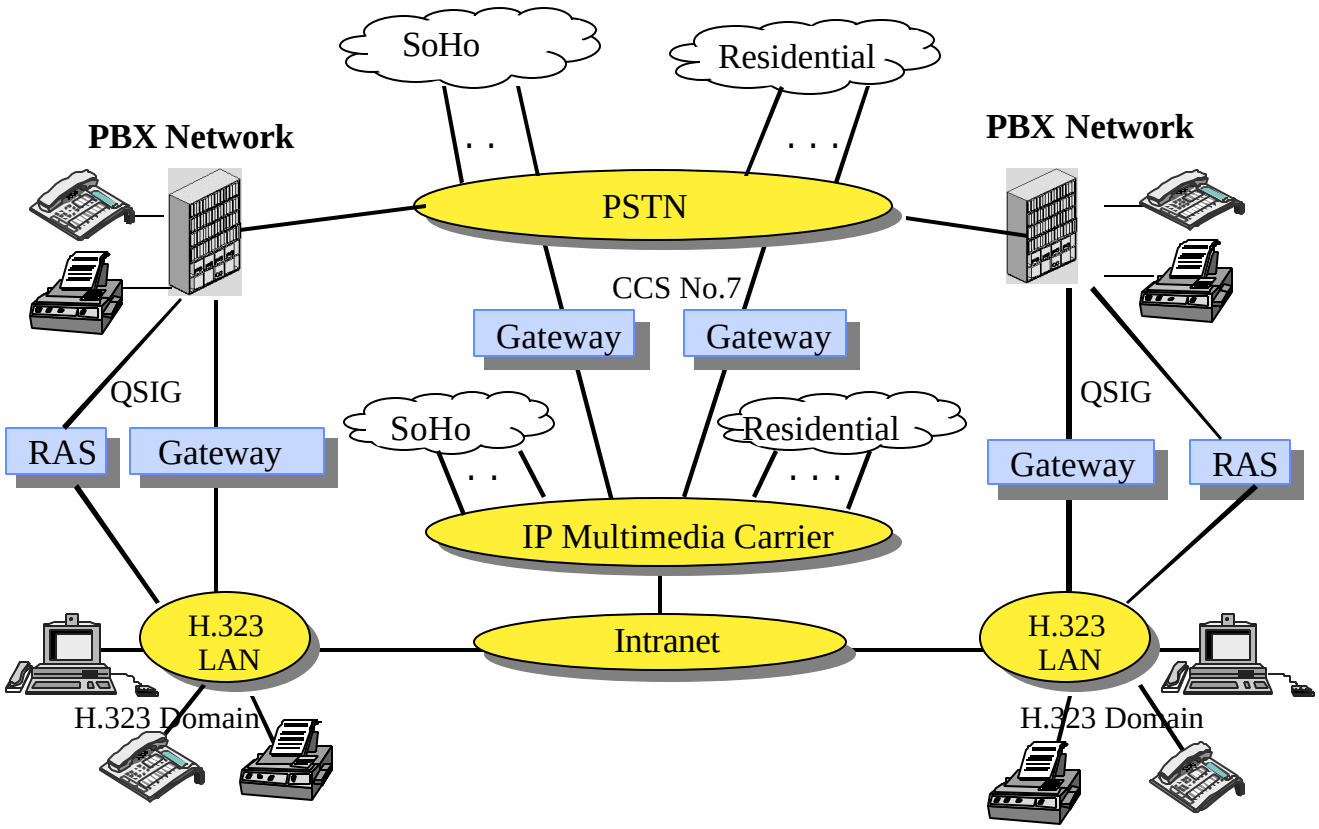
With the emergence of multimedia on packet networks, traditional telephone services are being developed for H.323 IP multimedia telephony networks. However, the architecture for signaling is different from the traditional centralized switch model in voice telephony. This paper describes the architecture and design of supplementary services in H.323 and compares the architecture to that of the switch model. The architectural model is peer-to-peer, the protocol design is based on the QSIG standards, and the services can be designed using a multi-tier approach. The peer-to-peer model reduces the dependence on the network to routing the signals and data; this is in sharp contrast to the traditional telephony switch model. QSIG is available as a worldwide standard (ISO/IEC JTC1) for private integrated services digital network (ISDN) telecommunication networks. Using the QSIG protocol design reduces the complexity involved in inter-working with the traditional circuit-switched equipment that uses QSIG. The multi-tier approach of designing services allows for complex services to be easily developed using building blocks consisting of simple services.

INTRODUCTION

The ITU-T H.323 [1, 2] series of recommendations describes terminals, equipment, and services for multimedia communication over packet-based networks (e.g., IP networks), and it covers the protocols necessary for their operation and for inter-connection with circuit-switched networks. H.323 terminals and equipment may carry real-time voice, data, and video, in any combination.

An example of the H.323 multimedia network is shown in Figure 1 where the H.323 network is interconnected to a legacy circuit-switched network.

Figure 1 shows some aspects of both the carrier and the corporate network along with the legacy circuit-switched network. Native H.323 calls for voice, facsimile, and other multimedia can be made within the H.323 domain; between H.323 domains using the intranet; within the H.323 domain in a carrier network; between subscribers of the H.323 domain in a carrier network and a corporate network; and between virtual private network (VPN) subscribers of the H.323 domain in a carrier network (Virtual Centrex Service). Inter-network H.323 calls can be made between PBX extensions and corporate H.323 users via the QSIG gateway; between PBX extensions belonging to different PBX networks via the H.323 networks for the purpose of toll by-pass; between residential public switched telephone network (PSTN) subscribers and corporate H.323 users via the common channel signaling system No. 7 (CCS No. 7) gateway; between PSTN and multimedia services network via the CCS No. 7 gateway; and between two PSTN subscribers via the multimedia services network using CCS No. 7 gateways for the purpose of toll by-pass.



Legend: SoHo = Small-office / Home-office
RAS = Remote Access Service
CCS No.7 = Common Channel Signaling System No. 7
QSIG = D-Channel signaling protocol at Q reference point for PBX networking

Figure 1: Example of an H.323 network interconnected to the legacy circuit-switched network

The supplementary services are covered in the H.450 [3] series of recommendations. In this paper, the architecture of supplementary services is described. Since the architecture is sharply different from that of circuit-switched networks, comparisons are made whenever appropriate.

ARCHITECTURE

The key elements of H.450 supplementary services are protocol based on ISO QSIG, peer-to-peer signaling, and a multi-tier approach to developing services.

H.450 Based on ISO QSIG

As services in the multimedia networks are deployed, they will have to inter-work with the services in the circuit-switched installed base. Being based on QSIG enables simple inter-working with the widely deployed installed base of private telecommunication networks that are QSIG based. Also, being based on QSIG enables smooth migration from the current installed PBX networks to H.323 multimedia networks.

H.450 Based on Peer-to-Peer Signaling

The H.450-based peer-to-peer signaling is different from the third party signaling of the switch. This difference results in changes to the way services are deployed, provisioned, charged, and to how incompatibility is resolved.

Service Deployment

The H.450 model is like the Internet model where intelligence resides at the ends/edges and the network simply routes the packets. The end device can be a PC or any IP phone, and the edge device is a PBX or a consumer gateway that resides at home. The state of the calls (ringing, busy, waiting, held, etc.) is also distributed to the ends/edges. This is in sharp contrast to the traditional switch model where intelligence and state of all calls is in

the network, and the ends/edges are simple phones running a stimulus-response protocol.

In H.450, services are deployed in much the same way as any other software package that is purchased from a store or downloaded from a Web site and installed on the end/edge device. Services can be developed by any vendor and sold directly to the end-user for deployment. In this environment, it is most likely that a huge industry will develop that will result in tremendous innovation in creating new services. In the switch model, services are deployed at the switch, which may result in other parts of the network being upgraded by the service provider before it is made available to the user. This involves a huge up-front cost for the service provider before any return on investment can be achieved from the users of the service. However, deploying services in a centralized location reduces some complexity as compared with the distributed nature of deployment in H.450.

Service Provisioning

In H.450, services are provisioned automatically by running the software package containing the services. In the switch model, services are provisioned by the service provider at the switches and obtained by the users through the use of DTMF and ringing tones.

Service Charging

Due to differences in the deployment of services, the model for charging is also different. In the H.450 model, the user pays for unlimited use of the services software up front. In the switch model, the service provider usually charges the user on a monthly basis. In H.450, the service provider has the option of imposing a small charge on the user for using the network for signaling, if the signaling is routed and monitored via the service provider's gatekeeper.

Service Incompatibility

In H.450, incompatibility of services between clients is resolved through the use of capabilities. The clients exchange their capabilities, and the services that are within the capabilities of both the clients can be executed. In the switch model, the more capable switch executes services on behalf of the less capable switch.

Multi-Tier Approach to Develop Powerful Services

One of the strengths of H.450 is that it enables a multi-tier approach to the development of services. Basic services consist of building blocks or primitives from which more complex services are developed. Compound services are developed from two or more basic services. Both basic and compound services are used by applications to provide multimedia services to the user. Examples of basic services supported in H.323 include multiple call handling, call transfer, call forwarding, call park and pickup, call waiting, message waiting indication, and n-way conference. Examples of compound services include consultation transfer and conference out of consultation. Consultation transfer uses call hold, multiple calls, and call transfer. Conference out of consultation uses call hold, multiple calls, and n-way conference.

BASIC SERVICES

The basic services that have been standardized in H.323/H.450 [3] are briefly described below.

Multiple Call Handling

This service [1] allows a multimedia client to handle multiple calls simultaneously.

Call Transfer

This service [4] enables the served user A to transform an existing call (user A to user B) into a new call between user B and user C selected by user A. User A may or may not have a call established with user C prior to Call Transfer.

Call Forwarding

This service [5] is also known as Call Diversion and it comprises these services: Call Forwarding Unconditional, Call Forwarding Busy, Call Forwarding No Reply, and Call Deflection. It is applied during call establishment, and it diverts an incoming call to another destination alias (e.g., telephone number, IP address, e-mail address) address.

Call Forwarding Unconditional permits a served user to have incoming calls, which are addressed to the served user's number, redirected to another number.

Call Forwarding Busy enables a served user to have calls, which are addressed to the served user's number and meet busy, redirected to another number.

Call Forwarding No Reply enables a served user to have calls, which are addressed to the served user's number and for which the connection is not established within a defined period of time, redirected to another number.

Call Deflection permits a served user to respond to an incoming call offered by the served client by requesting diversion of that call to another number specified in the response. This request is only allowed before the called user has answered the call.

Applications can be developed to allow the above variants of call forwarding to operate on all calls, or only on selective calls that fulfill specific programmed

conditions or conditions manually entered by the user. For example, forwarding can be made dependent on various conditions such as the state of the called party, busy, no-reply, absent; the caller identification; the time of the day; the day of the week; etc. For each scenario, the user can program the forwarding of incoming calls to different destination addresses. For example, a user can program his/her client to forward all incoming calls between 8 a.m. and 5 p.m. on weekdays to his/her office phone, calls on weekends from specific Caller IDs to ring on his/her home phone, and all other calls to be forwarded to his/her voice mail. Programming of the destination address can be done locally at the home client or by remote programming via a connection to the home client. Such a rich service is not easily available in traditional telephony.

Call Hold

This service [6] enables the served (Holding) user A to put user B (with whom user A has an active call) into a hold condition (Held User) and subsequently to retrieve that user B again. During this hold condition, user B may be provided with music and/or video on hold. The served (Holding) user A may perform other actions while user B is being held, e.g., consult with another user C.

Call Park and Pickup

Call Park [7] is a service that enables the parking user A to place an existing call with user B (Parked User) to a parking position. The call can later be picked up by retrieving the parked party from the same terminal where the park took place or from another terminal. Call Pickup is a service that enables the picking-up user to pick up a parked call. Upon successful invocation of Pickup, the picking-up user is connected with the parked user B.

Services such as Call Park/Pickup are used in the Automatic Call Distribution environment where calls are not directed to a specific user/client but to, for example, the first available agent with some specific skill. Such services are best implemented in the Feature Server, which then interfaces with a group of clients.

Call Waiting

This service [8] permits a served user who is busy with one or more calls to be informed of an incoming call with an indication. The user then has the choice of accepting, rejecting, or ignoring the waiting call. The user calling the busy party is informed about the call waiting condition.

Message Waiting Indication

This service [9] provides a general-purpose mechanism by which a user can be advised that messages intended for that user are available. A variety of message types are supported, such as voice mail, fax, and telex. In one of its simplest forms, when a message is left for a user, a Message Center sends a notification to the Served User, where a Message Waiting LED is lit. Additional information provided by the notification mechanism allows the Served User to know the number of messages that are waiting, the types of messages, the subjects of the messages, and the priority of the highest priority message.

N-Way Conference

This service [1] allows a multi-party conference to be established. This can happen as a result of two or more simultaneous calls merged into one conference or as a result of an initial two-party call later expanded to a conference. The limit on the number of participants in a conference is usually based on the policy of the entity hosting the N-Way Conference.

COMPOUND SERVICES

Compound services, on the other hand, use basic services to provide more powerful services to the end user. Two such services, Consultation Transfer and Conference Out of Consultation, are described below in detail.

Consultation Transfer

In Consultation Transfer, the user can perform three operations: (1) put a multimedia call on hold and retrieve it later, (2) call another person and optionally alternate between the two calls, or (3) transfer the call. The operations involve three supplementary services: Call Hold, Multiple Call Handling, and Call Transfer.

Call Hold. During consultation, the first call between A and B needs to be held so that it does not interfere with the second conversation between A and C. This is shown in Figure 2 In this case, the holding client A stops sending multimedia information (i.e., voice, video, and data) to the held client B so that the held party B cannot hear or see what happens during the consultation. At the same time, the holding party A is not able to see or hear the held party B. The simplest form of hold is called *near-end Hold*. In this case, the holding client A stops sending media, causing silence and “frozen picture” on the held client B. The other form of hold is *remote-end Hold*. In this case, the holding client A sends multimedia information, such as commercials, to the held client B (often called “music/video/document on hold”). At the same time, the holding client A may place the first call on mute and stop receiving multimedia packets from the held client B. The holding client A informs the held client B by sending a notification message via the signaling path.

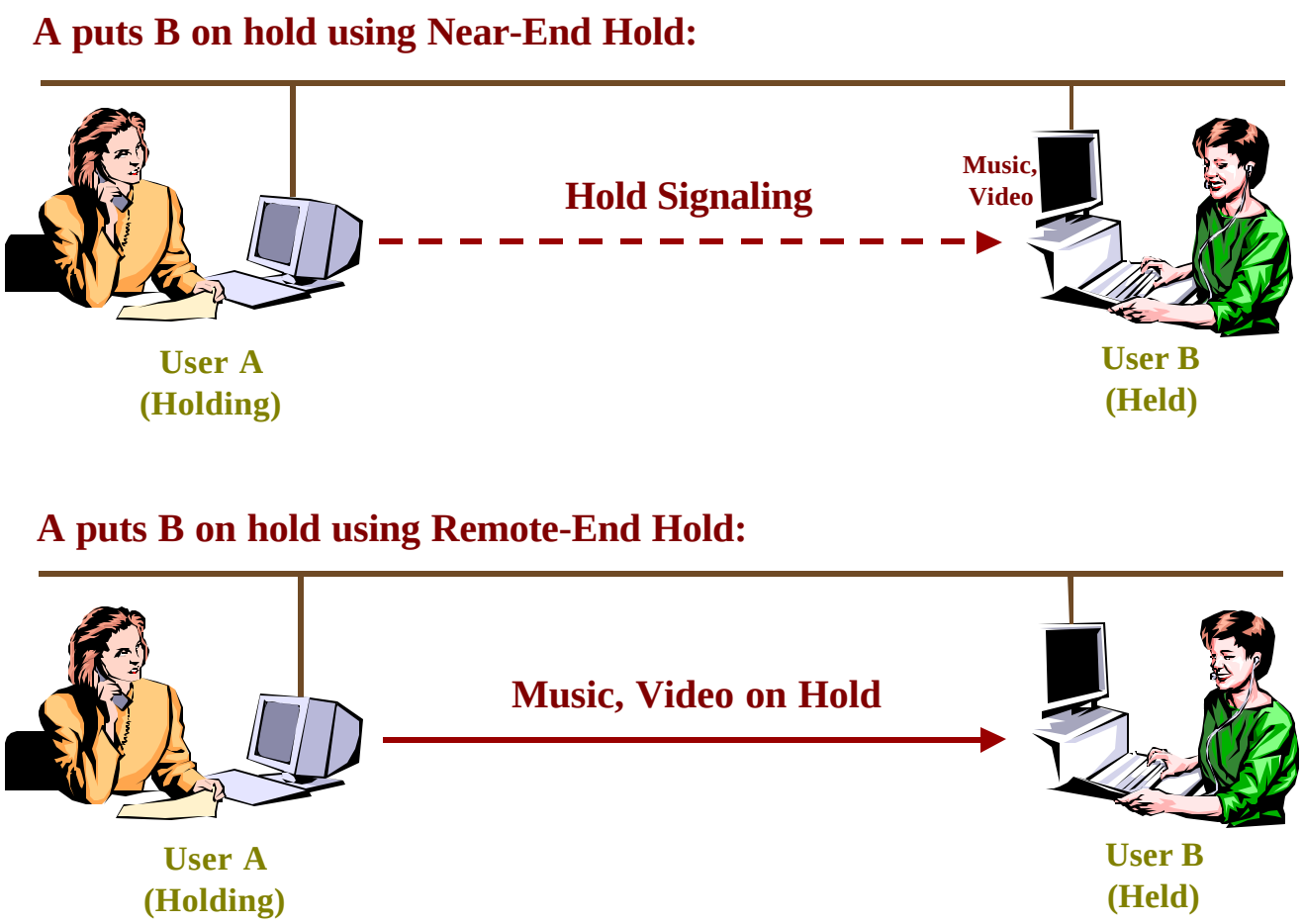


Figure 2: Call hold supplementary service

Multiple Call Handling. While the first call between A and B is held, the initiating client A establishes a second call to client C of the consulted party as shown in Figure 3. Since there is no hard limit on the number of H.323 calls a client can make simultaneously, this second call can be established in the same manner as the first call; in other words, it can be established as a basic call. Since party A is in two calls, one being held and another being active, he/she may swap between these two calls by putting the active call on hold and retrieving a previously held call.

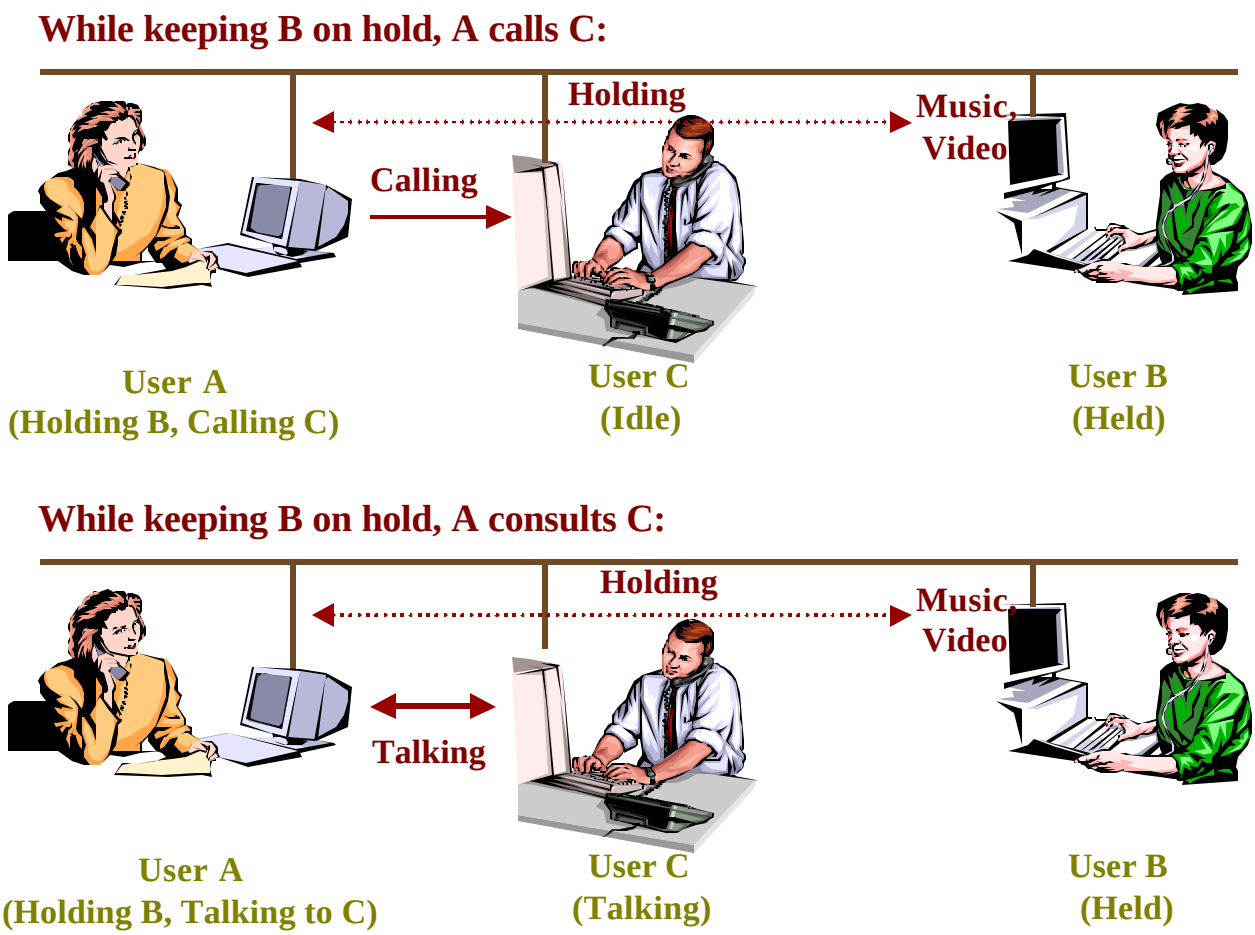


Figure 3: Multiple call handling supplementary service

Call Transfer. Call transfer is a basic service that enables party A to transfer an existing call with party C from his/her client A to client B as shown in Figure 4. Usually a multimedia conversation precedes the transfer, but it is not necessary. This transfer is performed in a single-step, i.e., the transferring client A sends a signaling message to the transferred client C requesting a new connection to be established directly from client C to the transferred-to client B. The transferred client C then establishes an independent connection to the transferred-to Client B. The first call between clients A and B is dropped if the transfer is successful.

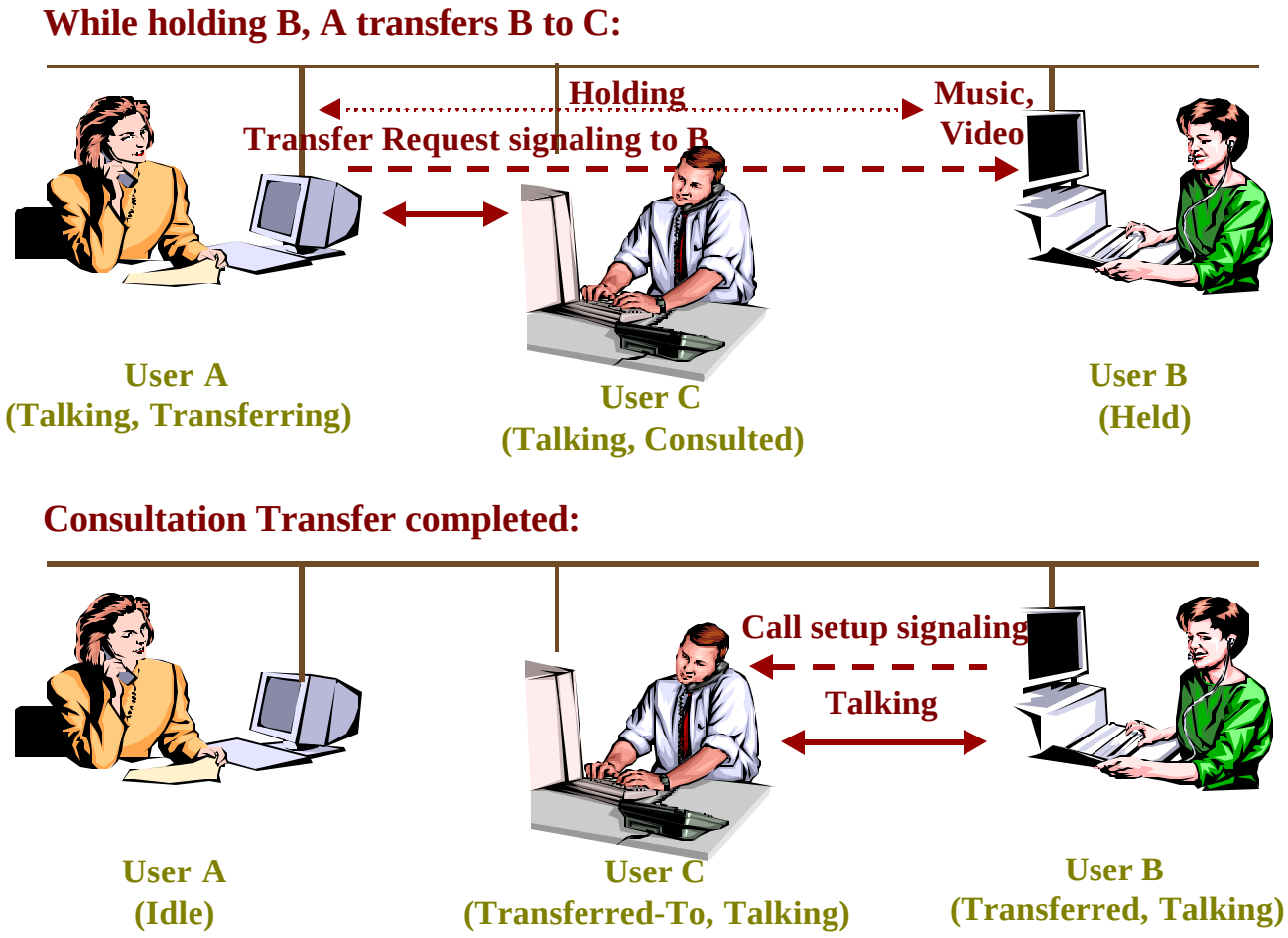


Figure 4:Call transfer supplementary service

CONFERENCE OUT OF CONSULTATION

In Conference Out of Consultation, the user can perform three operations: (1) put a multimedia call on hold and retrieve it later, (2) call another person and optionally alternate between the two calls, or (3) merge the calls in one conference call. The operations involve three supplementary services: Call Hold, Multiple Call Handling, and N-Way Conference. The services involving Hold, and Multiple Calls were discussed in the Consultation Transfer scenario where the call between client A and B was on hold and the call between A and C was active. The third service is described below.

N-Way Conference. This is a basic service that enables party A to merge the two existing calls between A and B and A and C into one conference call between A, B, and C. The signaling consists of client A transferring the calls between B and C to the Conference Server, and then making his/her own call to the Conference Server as shown in Figure 5. Previous calls between A and B and A and C are dropped if the conference is successfully established. In H.323, the address of the Conference Server is declared by each client as part of its capability during call setup time. The Conference Server could be resident in the client or be a resource on the network.

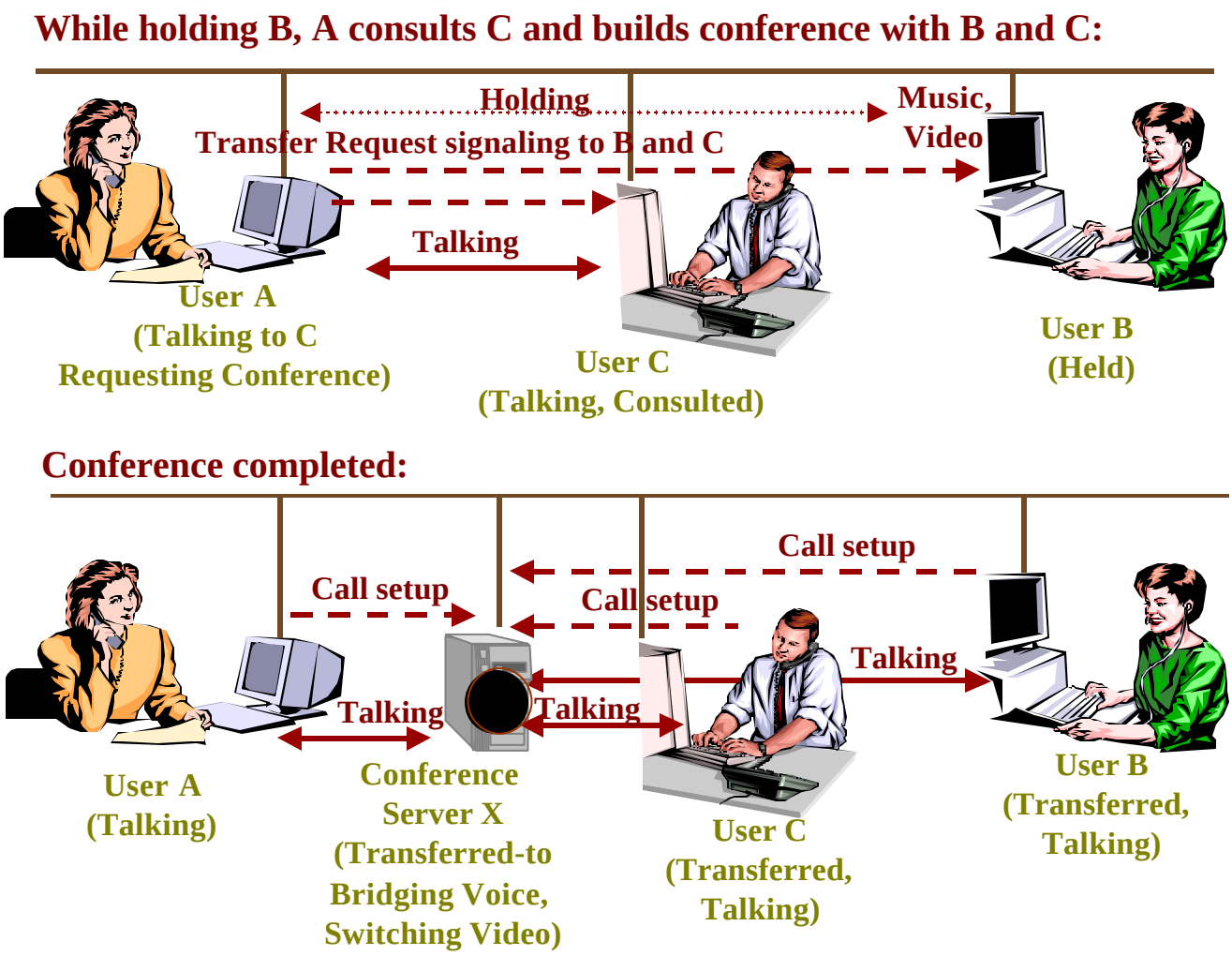


Figure 5: N-way conference supplementary service

CONCLUSION

International standards have been developed for deploying services on packet networks that rival services offered through circuit-switched networks. The architecture is suitable for use by both corporate networks and new-generation telcos. The deployment of H.323 has already begun in corporate networks. Savings in operational and upgrade costs are compelling reasons for unifying the voice and data networks. Unified networks provide richer services such as unified messaging involving multimedia instead of having voice-mail in the voice networks and e-mail in the data networks.

Even though the H.450 basic services developed so far mimic those already available in the circuit-switched corporate environment, the power of H.450 is in the multi-tier approach of developing services and in the distribution of service logic. More powerful and innovative services can be developed simply by using the basic services as building blocks. Since the service logic is distributed in the desktop phones as compared to being centralized at the PBX for the circuit-switched networks, a software PBX can be embedded in each phone. This removes the single point of failure at the centralized PBX and provides high scalability and fault tolerance in the H.323 network.

Since H.323 networks will coexist with the circuit-switched networks for a long period of time, using QSIG provides for smooth migration and simpler gateways to inter-work between the two networks.

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AUTHOR'S BIOGRAPHY

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