INTELLIGENCE IN SWITCHED AND PACKET NETWORKS

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The paper investigates the interaction between intelligent network (IN) and the Internet. Although there are many features of IN that can be recognized in the Internet, there is one important difference. In IN services are centrally controlled; in the Internet they are completely distributed through the network. If the Internet is used for telephony, there is a need of a protocol that sets up a call between computers or telephones. The interaction between IN and the Internet may be illustrated by Conference call service modelling. The paper considers Conference call service implementation in a public telephone switched network structured as IN and possible service models by the use of Session Initiation Protocol (SIP) and MEGACO protocol.

1. Introduction

The first step toward a feature-rich network was made with the introduction of intelligent networks. The main idea of IN is to separate call logic processing from switching, and to delegate it to specialized service control points (SCPs). IN standards are produced in the form of capability sets (CSs). The initial CS-1 and CS-2 were conceived for enhanced control of point-to-point voice calls. They are based on a call model that describes the states of setting up a voice call between two parties. IN was not originally intended for connectionless data services and does not deal well with conference type services involving multiple parties and multiple media streams. CS-3 and CS-4 were intended to address some of these features, but do not seem to live up to the expectations. New architectures may be necessary in the long term.

The Internet community influences the development of IN by specifying new protocols for Internet telephony, such as SIP, H.323, MEGACO, PINT, and SPIRITS. Although in a packet network there are no switching matrixes, there is still a need for nodes that terminate signalling protocols. The soft switches are nodes that terminate signalling and control network resources, but they are not necessary directly coupled to a switching matrix. The application of IN to soft switches is a hot topic.

In this paper some forces that Internet and IN communities exert on one another are examined. Models of a conference call service are presented to illustrate the different technologies.

2. Conference call model in an intelligent network

IN CS-1 deals with point-to-point telephone calls and does not offer sufficient features for multiparty and multimedia communications. CS-2 can handle calls between more than two parties. While in CS-1 the call and the connection setup as the result of the call were the same thing, in CS-2 it is possible to manipulate the individual parties in a call and their connections. One of the most important requirements is that the service logic must always have a consistent view of the configuration of the connections in a call. When the service logic loses track of who is connected to the call, the service gets into an inconsistent state and the effect of a join or split operation becomes unpredictable. For this reason CS-2 defines a formal model for describing the state of connections in a call. This is called the connection view state (CVS) model.

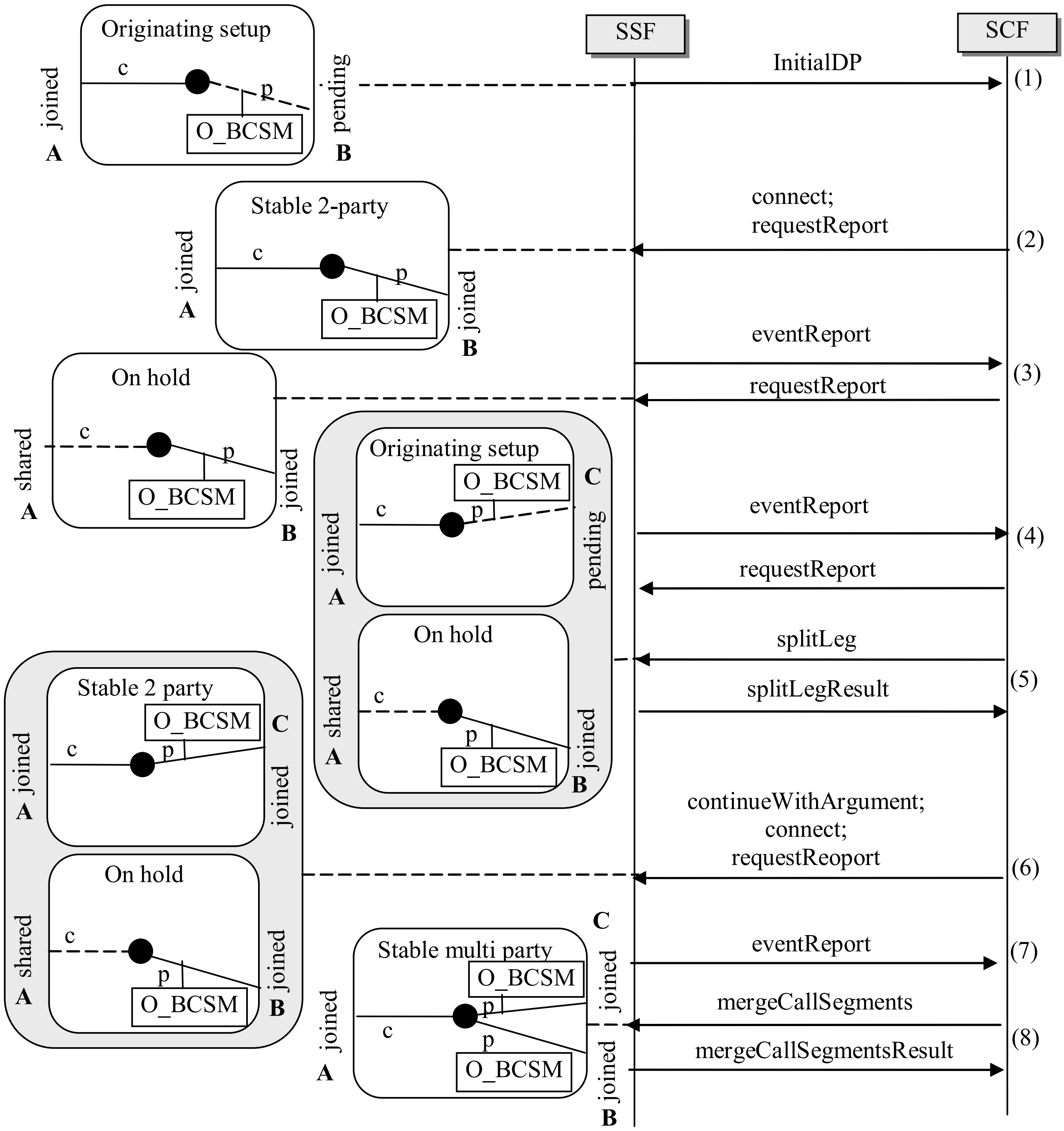

Figure 1 Connection-state view for a conference call

The main elements in CVS are *call segment* describing a half call in terms of legs and connection points and *call segment association* that is a group of all call segments that are involved in a given service.

Figure 1 illustrates an example of conference call between three users – A, B, and C. The initiator of the call is user A. The figure shows the dialog between the service switching function (SSF) that takes care of controlling the bearer connections in a call, and the service control function (SCF) containing service logic.

1. When the user A places the first call to B, the O_BCSM in SSF detects trigger and sends the *initialDP* message to the SCF.
2. The SCF instructs SSF to connect the originating and destination parties, and requests the SSF to report any midcall events.
3. When the user A puts on hold the current connection with user B, the SSF reports the midcall event to the SCF.
4. When the user A dials the number of user C, the SSF reports the midcall event to the SCF.
5. The SCF requests the SSF to separate one party from its call segment and place it in a new call segment.
6. The SCF instructs SSF to connect the originating party A to the third party C.
7. The user A flashes the hook to connect both parties.
8. The SCF requests the SSF to merge two associate call segments into a single call segment. It reestablishes the bearer connection between all involved legs. Users A, B and C are involved in a conference call.

Of course, the example is simplified; the messages between the SSF and SCF are more complex in reality. The focus is on CVS that represents how SCF sees the connections in SSF. When a command from the SCF to the SSF changes the connections in a call, the CVS changes accordingly.

CVSs can be very complex because IN CS-2 tries to implement multiparty communications by using classic switching technology intended only for point-to-point telephone calls. CS-3 is little more than revised version of CS-2. CS-4 was to accommodate the voice, video, multimedia, data communications, possibly involving multiple parties and using packet-based transmission rather than switched connections. CS-4 introduces the session manager (SM) handling voice and multimedia calls over Internet using either of the two well-known protocols H.323 or SIP.

Figure 2 gives a simplified view of SM in CS-4 as it omits some functional entities already defined in CS-1, CS-2, and CS-3.

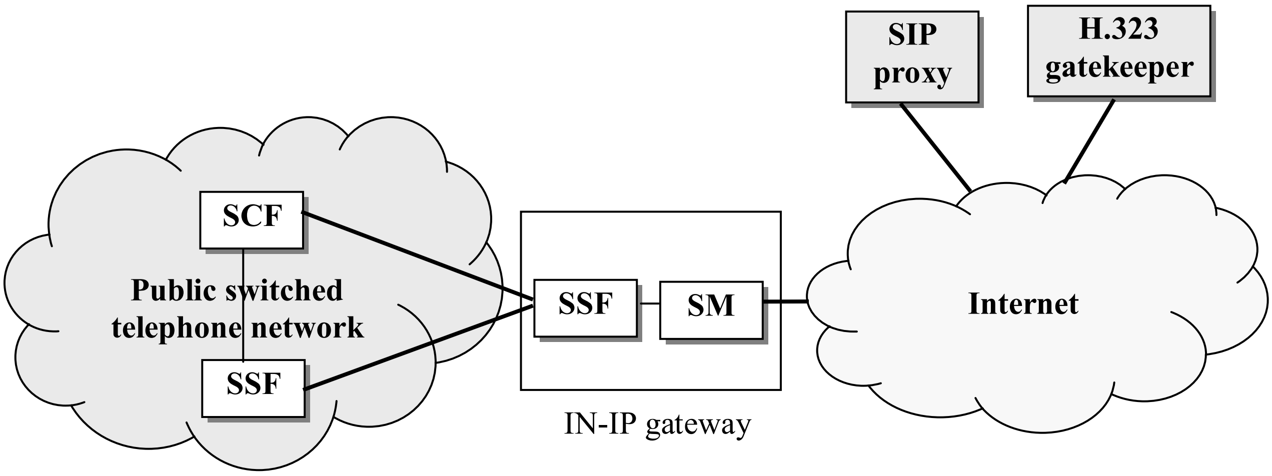

Figure 2 Partial view of CS-4 distributed functionality

The wide accepted by all standardization bodies distributed architecture for the Gateway implementation is based on three components – Media Gateway, Media Gateway Controller and Signaling Gateway. Figure 3 depicts the Gateway architecture distributed into the Call control functionality and the Media processing functionality over different network elements (a concept of soft switch).

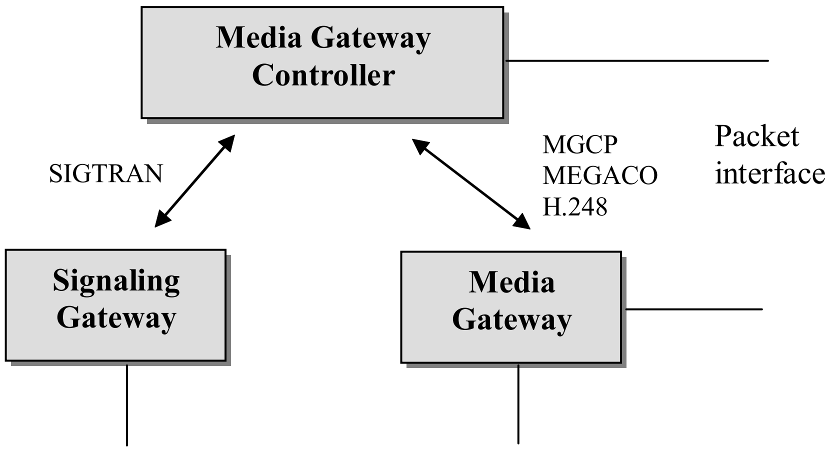

Figure 3 Gateway reference architecture

3. MEGACO connection model of a conference call

MEGACO is an application protocol for controlling Media Gateways from external call control elements called Media Gateway Controllers or call agents. A Gateway is typically a network element that provides conversion between the audio signals carried on telephone circuits and data packets carried over the Internet or over other packet networks. The foundation for the specification of MEGACO is a connection model that describes terminations and context. There is a striking resemblance between the connection model for MEGACO and the connection view state in IN CS-2.

A termination is a point on the media gateway that receives or sends media streams. A termination may be a physical port such as telephony line, but it can also be a media stream of temporary nature, such as RTP stream. A context describes how media flows between terminations in the media gateway. Contexts and terminations can be manipulated with MEGACO commands.

Figure 4 illustrates a MEGACO connection model of the conference call service.

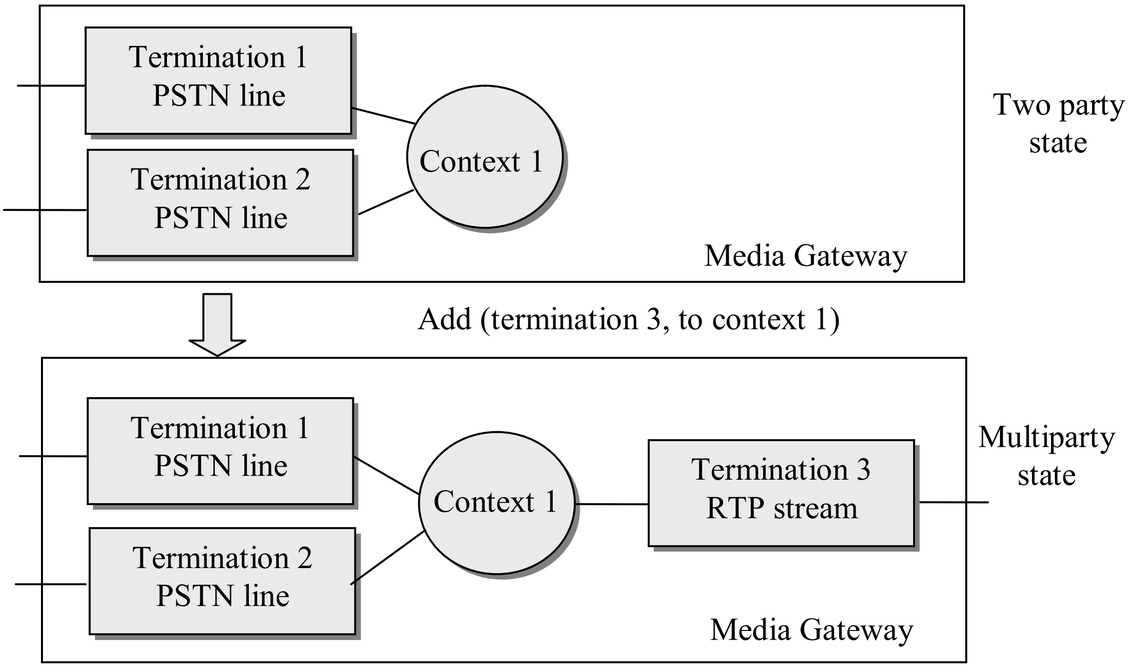

Figure 4 Example of terminations and a context for call conferencing scenario

In the top configuration termination 1 is connected to termination 2. In other words, the users connected to public switched telephone network (PSTN) are communicating. In this case the terminations are physical ports. Adding a packet data network subscriber in a conference is not different than adding a PSTN subscriber to a conference. The termination 3 is an ephemeral termination representing RTP stream. The *Add* command is used to create the termination 3 and to add it to the existing context. The concept of ephemeral termination brings about uniformity in representation in sense that the RTP stream can be operated upon in a manner similar to any other termination in the Media Gateway.

4. Conference call model in a SIP network

SIP is an application protocol for setting up multimedia sessions over the Internet. It is seen as the future of call signalling and telephony. One of the reasons for the rapid acceptance of SIP is that it is incredibly powerful call control protocol. It allows intelligent end points to implement the entire suit of telephony services without a service provider, and without a controller or switch for example. The signalling model for the conference call service in SIP telephony network is showed in the Figure 5.

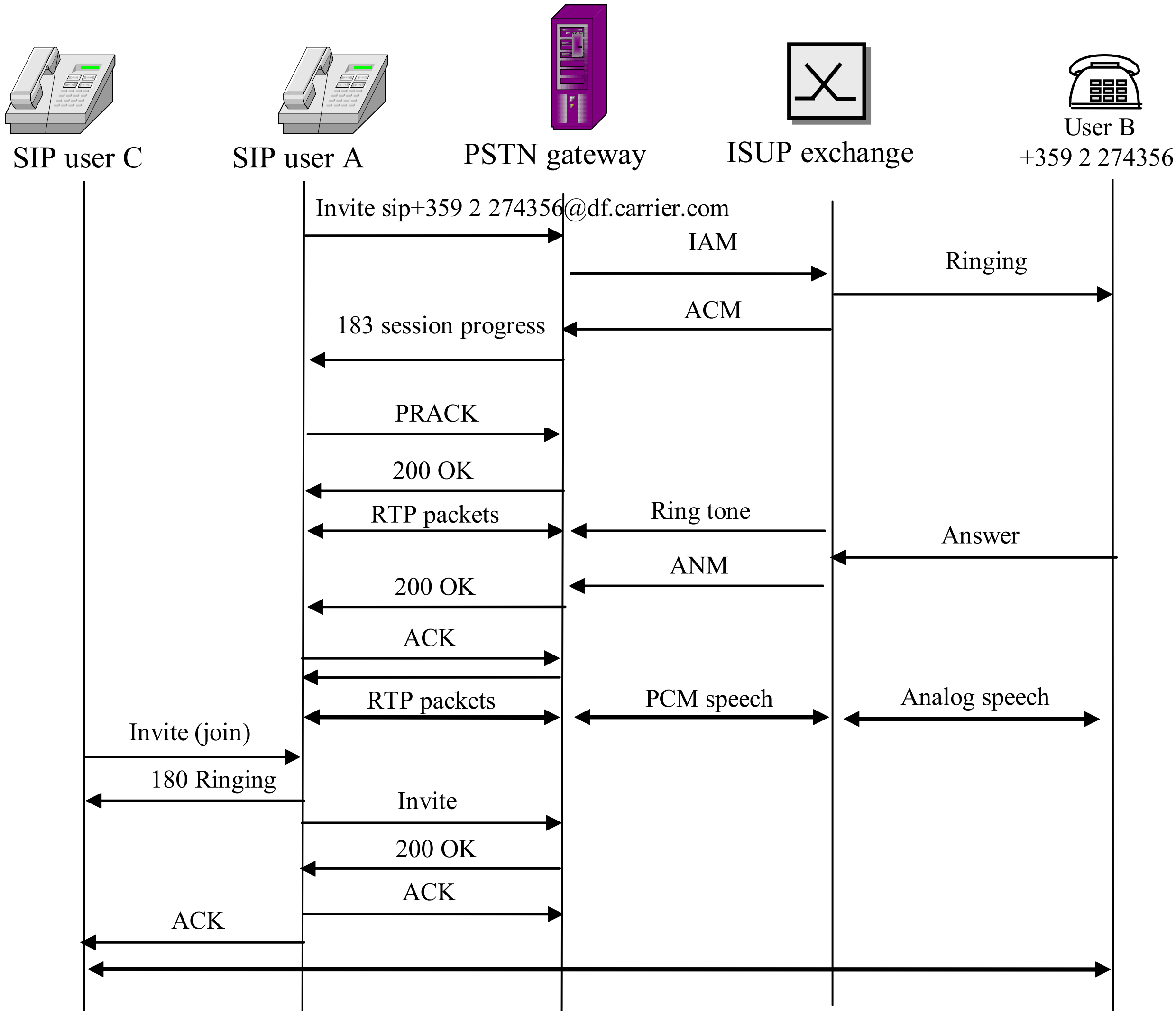

Figure 5 A signalling model of a conference call

In the example shown in Figure 5 the calling SIP phone **A** places a telephone call to the PSTN through a PSTN gateway. The SIP phone **A** has been preconfigured with the IP address of the PSTN gateway, so it is able to send it *Invite* request. The gateway initiates the call into PSTN by selecting an SS7 ISUP trunk to the next telephone switch in the PSTN. The dialed digits from the *Invite* are mapped into ISUP *IAM*. The ISUP *ACM* is sent back by PSTN to indicate that the trunk has been seized. Progress tones are generated in the one-way audio path established in the PSTN. Ring tone is generated by the far end telephone switch. The gateway maps the *ACM* to *183 session progress* response containing the RTP port that the gateway will bridge the audio from the PSTN. Upon reception of *183*, the caller's user agent begins receiving RTP packets sent from the gateway and presents the audio to the caller so they know that the call is progressing in the PSTN. The call completes when the called party answers the telephone, which causes the telephone switch to send *ANM* to the gateway. The gateway then cuts the PSTN audio connection through in directions and sends a *200 OK* response to the caller. Caller's user agent sends *ACK* to complete the SIP signaling exchange. SIP caller **A** and PSTN subscriber **B** are involved in a dialog.

Another SIP user **C** wants to join to the dialog established. His SIP phone sends *Invite* request with *Join* header field. *Invite (Join)* requests that the dialog be joined with an existing dialog. As the dialog is already part of a conference, the *Join* header field is a request to be added into the conference.

5. Conclusion

IN was born in an era when switched telephony networks ruled the world and the Internet protocol was still confined to the interconnection of data networks. But the world has changed. The Internet has become a general public utility. It is now perfectly possible to encode a voice conversation in packets and send these over an IP network. Today we faced with the challenge of creating value-added services in heterogeneous networks with both circuit-switched and packet-based transmission.

A profound analysis shows that services traditionally offered in telephone networks may be also provided in IP networks. New protocols for Internet telephony such as SIP, H.323, and MEGACO contribute to combination of Internet applications and telephony services. The paper examines some ways of implementing IN services in Internet.

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