

Implementing Intelligent Network Services with the Session Initiation Protocol

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Abstract

Internet telephony is receiving increasing interest as an alternative to traditional telephone networks. This article shows how the IETF's Session Initiation Protocol (SIP) can be used to perform the services of traditional Intelligent Network protocols, as well as additional services.

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1 Introduction

In the development of Internet telephony, we want to ensure that all the features supported by modern advanced telephony systems can be supported. This article describes many of the features as they are implemented in traditional telephone networks, and then describes how they can be implemented in Internet Telephony with the IETF's Session Initiation Protocol and its extensions.

The initial task of enumerating a large number of advanced telephony services is the same one that Study Group 11 of the International Telecommunications Union Telecommunications Standards Sector (ITU-T) addressed in the process of developing their standards for Intelligent Networks. The study group published its accumulated descriptions of services and service features in Annex B of ITU-T recommendation *Q.1211: Introduction to Intelligent Network Capability Set 1* [1]. Since these service descriptions were compiled from a number of disparate sources, the document acknowledges that they may be self- and mutually-inconsistent.

This paper will describe the implementation of Internet Telephony services by following Q.1211's descriptions of each service and service feature, noting the specific description of each one so as to clarify the ambiguities and inconsistencies in the descriptions.

Study Group 11 has written a follow-up document, *Q.1221: Introduction to Intelligent Network Capability Set 2*. This document has not yet been formally ratified or released by the ITU; I address it, in less exhaustive detail, in section 6.

2 Overview

The architectural model of Internet telephony is rather different than that of the traditional telephone network. The base assumption is that all signaling and media flow over an IP-based network, either the public Internet or various intranets. This is a dramatic change in the ability of nodes in the network to communicate: in the traditional telephone architecture, nodes can generally only communicate with those other nodes to which they are directly connected.¹ IP-based networks, on the other hand, present the appearance at the network level that any machine can communicate directly with any other, unless the network specifically restricts them from doing so, through such means as firewalls.

This architectural change necessitates a dramatic transformation in the architectural assumptions of traditional telephone networks. In particular, whereas in a traditional network a large amount of administrative control, such as call-volume limitation, implicitly resides at every switch, and thus additional controls can easily be added there without much architectural change, in an Internet environment an administrative point of control must be explicitly engineered into a network, as in a firewall; otherwise end systems can simply bypass any device which attempts to restrict their behavior.

In addition, the Internet model transforms the locations at which many services are performed. In general, end systems are assumed to be much more intelligent than in the traditional telephone model; thus, many services which traditionally had to reside within the network can be moved out to the edges, without requiring any explicit support for them within the network. Other services can be performed by widely separated specialized servers which result in call setup information traversing paths which might be extremely indirect when compared with the physical network's actual topology.

¹This is somewhat of an oversimplification for the modern SS7 backbone signalling network. Signals do not necessarily follow the same physical path as the trunks; however, except for some specialized functions such as lookups in the 800-number database, call-setup signals must proceed hop-by-hop from one switch to another, indirectly following the trunks' path.

Most of the services and service features of ITU-T Q.1211 can be provided by the IETF's draft signaling standards for Internet Telephony, SIP (the Session Initiation Protocol) [2] and, for some more specialized features, its Call Control extensions [3].

2.1 Billing

The one broad class of features which cannot be directly provided by SIP and SIP-CC is those which involve payment responsibility. In Internet telephony, it is still somewhat unclear what services can actually be charged for; clearly, for those services which can be performed by end systems, an external entity cannot expect to exact any fee for them, other than the one-time sales price the vendor of the end-system or its software receives. Those services which *do* reside in the network can be generally divided up into three categories: those residing at a single point, such as user-location or premium-rate end systems; those which involve better-than-best-effort packet delivery; and those which leave the Internet for some other network, such as PSTN gateways.

A number of billing models are possible for each of these types of services. Three of them seem to be most likely: the subscription model, the "New Jersey Turnpike" model, and the "Garden State Parkway" model. The subscription model involves a user paying for an unlimited amount of service in advance; this is most likely applicable for the single-point services discussed above. The "New Jersey Turnpike" model is payment by settlement — when a service is initiated, the user commits to paying after completion for however much service he or she has used. This is the model of current residential or charge-card service in traditional telephony. The third model, "Garden State Parkway", is pay as you go; a user commits some token before the service is initiated which allows a certain amount of service, and commits further tokens along the way. This is the model of traditional coin-operated telephone calls, or pre-paid calling cards.

Regardless of the billing model used, the format of billing information for Internet telephony is essentially orthogonal to questions of its signalling. Work is currently ongoing in the IETF's Internet Open Trading Protocol working group to define standards for exchange of such information.

3 Architecture of IPtel signaling

The flow and control model of SIP signaling is generally based on the existing models of Internet e-mail (operating at signaling rather than background-batch speeds) and (to a somewhat lesser extent) the World Wide Web.

Every SIP address specifies (through the usual DNS-style addressing) a server domain in which the address resides. Addressing is deliberately reminiscent of e-mail: sip:(user)@(domain) or sip:(user)@(host). An end system, when placing a call, will either directly look up the remote destination specified in the address, to send it the SIP invitation directly, or it will forward the call invitation to a local proxy, which will perform this lookup function for it (and can perform other functions as well). The resolved destination may be the address of the actual destination end system; it may be a recipient-side proxy, which handles such services as user location and firewall punch-through; or it could be a redirection server, which informs the originating station of a different server at which it should search for the user.

It is important to note that unless specifically architected (as in firewalls) no proxy server can guarantee that it will be on the server path for calls between any pair of end systems. Also note that the majority of the Internet path over which both media and signals will flow (the backbone) will never see any signaling information as anything other than IP packets to be routed to their destinations.

Some more specific details of the Internet telephony architecture are discussed in the sections describing their corresponding IN services.

4 Capability Set 1: Service Features

Q.1211 divides the services it describes into two broad categories: "services," which are what an Intelligent Network vendor would actually wish to provide to customers; and "service features," which are lower-level building blocks used to construct the services.

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