Internet Telephony: A Second Chance

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It is rare that one gets to re-engineer a fundamental part of the communications infrastructure that has existed for more than a hundred years. We now have this once-in-a-lifetime chance, but must resist the temptation to simply recreate the same old network, just using packets instead of circuits. Below, we give some examples of how to emphasize *Internet* telephony rather than Internet *telephony*. We also describe our perception where the open issues are and possible approaches.

First, this design principle implies that we should restrict any PSTN-specific features and assumptions limited to gateways and translation software, rather than making Internet devices aware of these legacy technologies. Such legacies include E.164 telephone numbers, voice-only orientation, the use of voice prompts or messages and in-band signaling such as DTMF.

Where sensible, services inspired by the PSTN should be made to work across all Internet services, not just telephony. For example, emergency call services ("911" in the US, 110/112 in Germany) should work the same way whether invoked from a chat tool, email or an Internet application. Locating gateways, as implemented in TRIP [1], [2], may be applicable as a wide-area service location protocol for both electronic and physical services. Also, dynamic carrier selection, available on a per-call basis in the PSTN, needs to be made available for QOS-controlled IP services.

In the long run, it is likely that Internet telephony service will not be a stand-alone offering, but rather be part of a large set of applications that do not look like a telephone at all. Internet phone calls will be initiated from chat applications, distributed games, virtual reality environments, web pages and applets embedded in email. SIP, for example, accommodates this, making it easy for web pages to contain SIP URLs for one-click-dialing and allowing SIP responses to contain web pages or redirect calls to any other URL, such as email, web page or chat. Indeed, it has been suggested that chat and Internet telephony are so closely related that it makes sense to use a single signaling protocol for both [3], [4]. In that model, text chat is just one of many possible session types, including traditional telephony, multi-player games or conferencing.

Much of the complexity of the current PSTN arises from its charging model. The PSTN charging model is neither sufficient nor necessary for the new environment, except when charging for gatewayed calls into the legacy PSTN. Already, advertising-supported phone calls are becoming popular, as the cost per impression of about 0.6 to 6 US cents approaches the cost of providing a minute of domestic service. For higher-quality and video services, bandwidth-based charging, independent of the application, needs to be developed, possibly based on congestion-adaptive pricing models [5] that make it possible to offer affordable high-quality video service at least during off-hours.

Providing assured quality-of-service is probably more of an administrative than a technical problem at this point. Voice service, in particular, is a good candidate for differentiated services [6], as traffic engineering is relatively straightforward. In particular, the aggregated model [7] where RSVP or similar resource reservation protocols provide admission control for traffic classes. Simple prioritization for voice packets works well as long as VoIP is the major QOS-assured traffic class.

Probably the major challenge faced by Internet telephony is moving from the current Internet reliability of about 99% or 99.5% to 99.999%, i.e., no more than five minutes of unavailability per year. This requires a different mindset, not just protocol and technology fixes, as upgrades have to be done while the system is running and every component has to be engineered to have a hot standby. A particular problem for Internet telephony is that it requires a large number of components, including gateways, proxy servers, DNS, DHCP and resource reservation, each subject to independent failures, thus, simplicity and re-use of core infrastructure services is needed.

Many system failures are caused by misconfiguration. Particularly for Internet telephones, devices need to be able to be bought, plugged into an Ethernet socket and then function, without any further manual intervention. Configuration using DHCP [8] or SLP [9] work well in single-provider LANs, but may not be sufficient where a single access infrastructure such as CATV is shared by multiple operators.

In a few years, most telephone devices will be wireless. Next-generation networks such as 3G will push IP closer to the end system, but it would simplify the overall architecture if a single mobility protocol can handle both the cases of discontinuous and continuous mobility [10], [11].

Unified messaging, combining email, fax and voicemail into a single user interface, will be made much easier with the use of IP-based delivery. For example, MGCP [12] or RTSP [13] can be used to generate voice prompts or record messages, then delivered via SMTP and retrieved via IMAP or POP.

Unlike traditional telephony, where services are only available in a few pre-canned varieties, Internet telephony offers the opportunity to have service providers, administrators and end users customize their telephony services. So far, mostly call filtering and routing have been made programmable [14], [15], [16], [17],. but there are opportunities for programming media interactions.

In summary, Internet telephony should be viewed as an opportunity mostly for system integration, not for inventing fundamentally new Internet architectures. It will succeed not by replicating the existing phone network (except in its reliability), but by creating an open platform for experimentation and creation of new services.

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