Voice Service Interworking for PSTN and IP Networks

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ABSTRACT This article presents an overview of the main technical problems to be addressed for the provision of interoperable services between IP telephony and the PSTN. The pivotal element of the solution resides in an interworking function. This function is typically implemented in a gateway whose requirements and behavior are here analyzed in terms of signaling and control protocols (control plane) as well as user data transfer (user plane). The presentation is structured around these two planes. The control plane defines the set of signaling protocols to be used in each networking context and the translation between them. Detailed scenarios illustrate the signal translation in the gateway allowing for the establishment of a hybrid phone call. The user plane is responsible for adapting the user data to the properties of each network channel and determines the quality of service of the voice call in terms of delay and speech quality.

Internet telephony is becoming a very successful voice technology, as evidenced by the burgeoning market for IP-based telephony products. This was enabled by recent advances in different technologies such as signal processing, switching, transmission, and processing power.

On the other hand, the public switched telephone network (PSTN) has made very impressive achievements in terms of coverage, reliability, and ease of use. Today, the number of lines is about to reach the milestone of 1 billion. The availability of the service is such that users are used to getting a dial tone every time they pick up the phone and to be connected to any selected called party. In addition, the telephone network is being extended by cellular networks, the growth of which is almost as dramatic as that of the Internet.

Matching these features with a fully IP-based network is a major engineering challenge that might last several decades; there is even no consensus today that this will ever happen. Some of the voice services so far offered by the PSTN will certainly migrate to an IP-based technology. However, we believe that IP telephony and PSTN services will coexist for a considerable time.

For these reasons, the ability to interconnect IP telephony users to PSTN users is an essential feature. It is the goal of this article to discuss the main interworking aspects between IP telephony and PSTN voice services.

Two main standardization approaches are being carried out for IP/PSTN interworking. In the IP world driven by the Internet Engineering Task Force (IETF), interworking with the PSTN has been the result of a logical extension to the IP telephony service, which is seen as one of many IP applications. AVT, IPTEL, MMUSIC, and PINT are the main IETF working groups concerned with IP telephony. In the telecommunications world, the International Telecommunication Union — Telecommunication Standardization Sector (ITU-T) has initiated various standardization activities (e.g., [1–4]) which captured the attention of most of the industrials involved in the field. Related to these standards, the European Telecommunications Standards Institute (ETSI) project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) undertook the effort to identify additional technical agreements required for interoperability between IP and circuit-switched networks [5]. Some industrial consortia such as the International Multimedia Teleconferencing Consortium (IMTC), through its Voice over IP (VoIP) group, also provide recommendations related to the implementations interoperability that is required in a multivendor context [6].

In this article we analyze the main requirements for interworking between IP telephony and PSTN services. Illustrations are based on the H.323 standard. For clarity, the interworking features are organized in two planes: the control plane and the user plane. Control plane interworking defines the set of signaling protocols to be used in each networking context and the translation between them. User plane interworking is responsible for adapting the voice data to the properties of each network channel and determines the quality of service (QoS) of the voice call in terms of delay and speech quality.

The article is organized as follows. The next section defines hybrid voice services and gives basic communication scenarios for IP/PSTN interworking. PSTN/integrated services digital network (ISDN) protocols and H.323 systems are briefly reviewed. The interworking features in the control plane are then described, and we discuss signaling adaptation, addressing, and media control functions. User plane interworking is discussed next. The impact of end systems and network design is analyzed in terms of speech quality and communication interactivity. The last section concludes the article.

VOICE SERVICE INTERWORKING

In this section we describe five basic scenarios for voice communications. We consider voice services over the PSTN and the IP network, as well as hybrid combinations. Also, we give a brief overview of the H.323 standard.

FIVE SCENARIOS FOR VOICE COMMUNICATIONS

Figure 1 illustrates five basic voice communication scenarios. Hybrid voice services are represented by scenarios 3, 4, and 5. In these scenarios, an interworking function (IWF) is needed to perform all protocol conversions and data adaptations. An IWF device may be used to connect two networks (i.e., a network adaptor) or a terminal to a network (i.e., a terminal adaptor).

For voice services, the IWF provides the following mechanisms:

- *Signaling adaptation:* This consists of the processing and translation of incoming signaling messages. It mainly concerns the call setup and clearing phases.
- *Media control:* This consists of identifying, processing, and translating service-specific control events that may be generated by the user or the terminal.
- *Media adaptation:* This consists of adapting the voice data to the data transfer channel of the downstream network.

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THE PSTN VOICE SERVICE

In scenario 1, two standard phone sets are connected via the PSTN. Although well known in the communications community, we briefly review the main PSTN characteristics that will be crucial to further discuss interworking concepts.

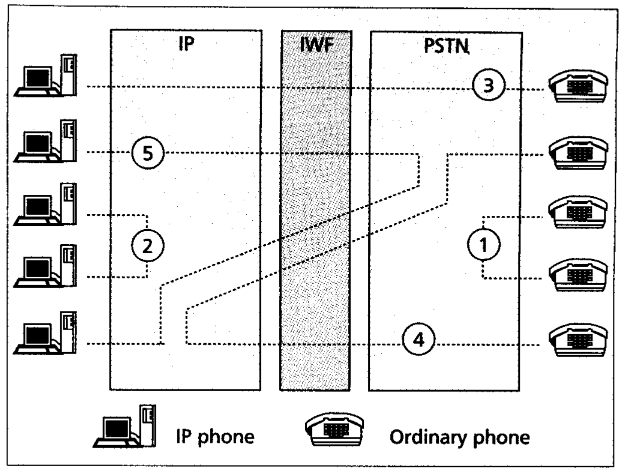
The PSTN core network is based on a circuit-switched network where each circuit corresponds to a 64 kb/s digital channel. A PSTN terminal can be either digital or analog. Standard phone sets are attached to the PSTN by means of an analog access network, which merely corresponds to the set of subscriber loops (the copper wires that link the customers to the central office). On an analog access network, voice is transmitted as a 3 kHz wideband analog signal and gets digitized at the access switch. In this case, signaling capabilities on the analog part of the access network (e.g., address notification) are reduced to in-band coding of dual-tone multifrequency (DTMF) tones.

The ISDN allows voice terminals to have digital access to the PSTN. In this case, a digital voice terminal (or an analog terminal attached to an adaptor) initiates a signaling dialog using Q.931 [7] (or Digital Subscriber Signaling System No. 1, DSS-1) to connect to the network via a 64 kb/s digital channel. Signaling inside the digital core network is based on Signaling System No. 7 (SS7) [8]. An ISDN terminal seamlessly calls an analog PSTN terminal and vice versa. A unified addressing system is defined in ITU-T Recommendation E.164.

Finally, one essential feature of the PSTN is its service creation and control capabilities referred to as *intelligent network* (IN) [9]. Basic services such as call forwarding rely on the IN architecture.

VOICE SERVICES OVER IP

Scenario 2 illustrates what is generally referred to as IP telephony. In IP telephony all service-specific processing and protocols, such as signaling and media coding, are pushed to the end systems and are transparent to the network. Part of the application may be built on top of TCP or UDP, depending on whether they are loss- or time-sensitive, respectively. For example, the TCP transport protocol is used to carry the signaling stream since the signaling channel has to be error-free. However, because of its intrinsic timing constraint, voice data is usually transmitted over UDP. The time-continuous property of voice signals requires that the transport channel ensure the appropriate streaming needed for data resynchronizing at the receiver.



■ **Figure 1.** *Voice communication scenarios.*

For this reason, the Real-Time Protocol (RTP) [10] is used. The sequence numbering field of RTP packet headers is used to reorder the receiving packets in case of out-of-sequence delivery (UDP does not ensure packet sequencing); the timestamp field indicates the temporal play-back position of the data payload.

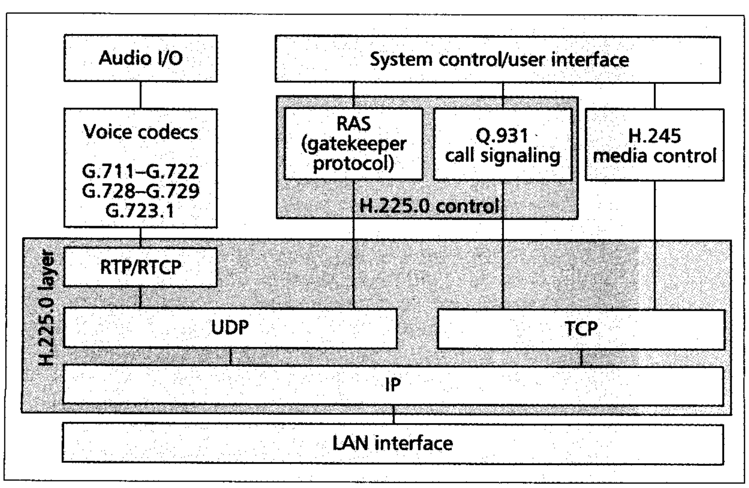
ITU-T Recommendation H.323 [1] and its related set of standards for packet-based multimedia communications [2–4] — in addition to several related efforts carried out by the ETSI, IETF, and IMTC — certainly constitute the most advanced framework to cover essential IP telephony issues. Although it is not our goal to present a tutorial on H.323, a brief description of the standard is required for the following discussion on interworking. The presentation is restricted to the basic voice aspects of H.323; data and video communications as well as multipoint aspects are not covered.

H.323 Systems — The H.323 standard defines three types of equipment: *gatekeepers*, *gateways*, and *terminals*. The *gatekeeper* is an optional equipment that provides call control services to the terminals. Examples of such services are address translation, admission control, call authorization and directory services. The RAS (Registration, Admission and Status) protocol defined in H.225.0 is used to communicate between a terminal and a gatekeeper.

The *gateway* is responsible for providing all translations necessary for transmission formats and control procedures between the IP supported portion and the PSTN/ISDN part of hybrid calls. Since gateway functions are more related to hybrid calls than pure IP calls, they will be discussed in the next section.

The *H.323 terminal* components are described in Fig. 2. A terminal may support several standards for voice coding. The G.711 codec (used in ISDN) is, however, mandatory for all terminals. H.225.0 specifies the use of logical channels based on the RTP/UDP/IP protocol stack to transfer coded voice data. The *system control part* of a terminal is composed of three protocols:

- The RAS signaling function is used for the dialog between a terminal and a gatekeeper. The associated channel, called the *RAS channel*, uses the UDP/IP protocol stack. A main function of the RAS channel is to allow the terminal to be attached to a gatekeeper by registering itself. Registration



■ **Figure 2.** *An H.323 voice terminal.*

basically results in an update of the gatekeeper's address translation table. This allows other terminals to locate the registered terminal and to determine its transport address in order to initiate a call signaling channel.

- The call signaling between two H.323 terminals is based on Q.931 messages. The call signaling channel uses a TCP/IP protocol stack. The call setup phase consists of sending a Setup message to the destination. The setup phase is considered successful upon reception of the Connect message from the called terminal. The next phase is the establishment of the H.245 channel.
- The H.245 protocol defines end-to-end control messages used for capability negotiation (e.g., the supported codecs), opening and closing of logical channels, flow control messages, and so on. The H.245 control channel is a reliable channel based on TCP.

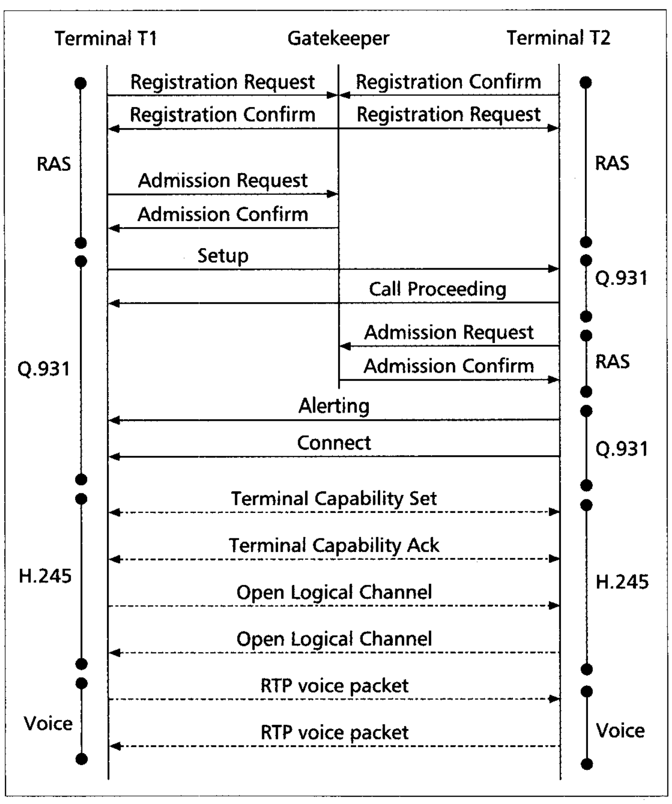
Figure 3 shows an example of a control protocol diagram between two H.323 terminals. A description of some of these messages is given in the third section. Finally, it should be noted that H.323 defines a Fast Connect method in order to alleviate the initiation phase in basic and simple calls. The H.245 dialog is then replaced by additional information elements in the Q.931 messages so that, upon reception of the Connect message, all needed voice channels are activated.

Hybrid Voice Services

In scenario 3, the two terminals involved in the call use different protocol stacks to communicate with their access networks. The protocol conversions occur at the networks' boundaries. Two terminals, of different types in this case, communicate with each other to ensure an ad hoc voice service to the end users. Scenario 3 requires the mapping both of media and media control channels, and between signaling protocols.

In scenarios 4 and 5 the same protocols are used at the interface of each terminal, but a different protocol is used in the backbone network.

The gateway is the equipment that generally hosts the IWFs. However, in the H.323 standard the gatekeeper may also be involved in some IWFs such as address translation. In the next sections we will generically call a *gateway* the equipment in charge of all IWFs. In addition, we will only consider the case where the PSTN/IP gateway is connected to the PSTN via ISDN access. Interworking issues between ISDN networks and analog PSTN terminals are not covered in this article.



■ **Figure 3.** *A diagram of H.323 control protocols.*

Signaling and Control

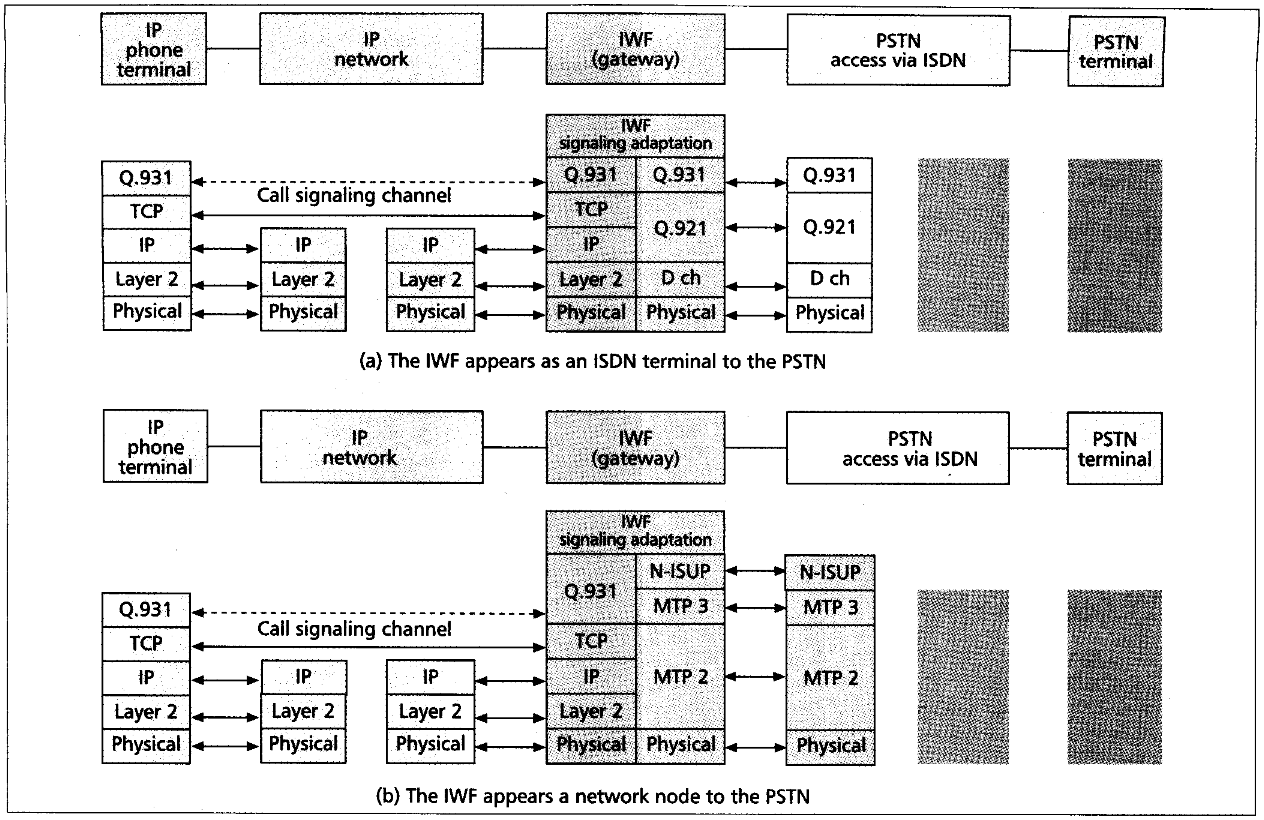
In this section we show how call connections are set up and control commands conveyed during a communication.

Signaling Adaptation Functions

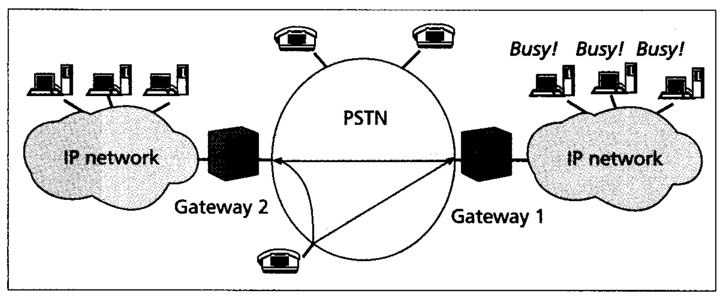
If two different signaling protocols are used in the interconnected networks, the IWF should translate the signaling messages in such a way that the end-to-end call can be completed. In the H.323 gateway, Q.931 is used in both the IP network and ISDN access. However, the Q.931 signaling channel between an IP terminal and the gateway is terminated at the gateway (i.e., Q.931 messages are processed in the gateway and not simply forwarded). A peer Q.931 channel is then used to support the call control on the PSTN side. This is mainly due to the fact that H.323 has defined a particular use of Q.931 messages, so there is not necessarily a perfect correspondence with the ISDN use of Q.931 [2, 3]. Figure 4a shows the IWF protocol stacks in the control plane in the case of scenario 3.

SS7 Interoperability

For historical reasons, IP/PSTN gateways are usually seen as administrative boundaries between a network provider (usually the operator) and a network customer (usually a company or an Internet service provider). For this reason, they are connected to the network as *terminals*. However, the gateway can be connected as a *network node* to the PSTN to have access to its SS7. Consider, for example, the scenario depicted in Fig. 5. Two IP-telephony based call centers are shown, each connected to the PSTN through gateways. The two call centers are combined to form a single virtual distributed call center; if all the agents in one call center are busy, the calls are to be diverted to the other one. If the gateways do not have access to the SS7 network of the PSTN, such call diversion requires terminating the call at the first gateway and reinitiating a call from the first gateway to the second one. This would tie up two PSTN ports of the first gateway, use up two voice circuits in the PSTN, and potentially introduce high delay due to the convoluted route the voice signal follows. On the other hand, if the first gateway has access to SS7, it can simply divert the call to be directly terminated at the second gateway, thereby avoiding the above inefficiencies. In this way, the two call centers can seamlessly be joined to form a virtual call center, which can be called at a common phone number. In this case, the gateway needs to implement the Narrowband ISDN User Part (N-ISUP) protocol. Figure 4b shows the protocol stack needed for scenario 3 with the gateway connected to the PSTN as an ISDN node.



■ **Figure 4.** *The IWF in the control plane.*



■ **Figure 5.** *SS7 interoperability: a call diversion scenario.*

SS7 is central to the operation of the PSTN. Therefore, telecommunications companies are very reluctant to expose their SS7 networks to gateway owners. A more acceptable approach is to provide SS7 access to a *signaling gateway*, which would control one or more *media gateways*. The signaling gateway would then reside on the premises of the telecommunications company, and communicate with the media gateways via a specific protocol. Work is in progress in the IETF and ITU-T Study Group 16 to standardize such a protocol. Several candidate protocols exist, such as the Media Gateway Control Protocol (MGCP). Another proposal uncer consideration is to use the H.323 signaling protocols for this purpose.

ADDRESSING

In the IP world, terminal addressing is generally based on alphanumerical streams whose resolution and directoring are based on hierarchically organized servers [11]. Similar addressing schemes for IP telephony are provided by the Session Initiation Protocol (SIP) [12] defined by the IETF. However, as a requirement of service interworking between the PSTN and IP, each PSTN user should be able to call an IP-attached user and vice versa. When the call is initiated from an IP terminal toward the PSTN, the E.164 destination address can easily be sent to the gateway and then across the PSTN. The problem is more complex when the caller is a PSTN terminal and the destination an IP terminal. This is partly due to the limited dialing capabilities of standard telephone sets, particularly if only an alphanumerical type of addressing is defined for the destination.

One of the crucial questions is whether the numerical expression of an IP address can explicitly be used in the identification of the IP terminal. An important requirement for service interworking is that the calling user be oblivious to the network (PSTN or IP network) to which the callee is attached. The ITU-T approach to solving this problem is to allow an H.323 terminal to be identified by several address aliases of different kinds, typically an E.164 address and an e-mail-like address [1]. Such an approach generally requires specific address translation, resolution, and registration services, which in H.323 are typically performed by the gatekeeper.

Figure 6 shows an example of a call control scenario with address resolution. A PSTN terminal initiates a call to an IP terminal using the E.164 address alias. The steps of this scenario are the following:

- The IP terminal registers with the gatekeeper by giving a network address, aliases of the network address, and the transport address of its signaling channel (i.e., the TCP port number and IP address). Examples of network address aliases are *user@host* and an E.164 address. The terminal sends as many Registration Request messages as necessary to register all its address aliases.

- The gateway registers with the gatekeeper in the same way.
- The gateway receives a Setup message from the ISDN access switch. This message contains the E.164 address of the calling PSTN/ISDN terminal and the E.164 address of the called IP terminal.
- The gateway sends back a Call Proceeding message to indicate that the call is being processed.
- The gateway sends a Location Request message to the gatekeeper asking for the channel signaling transport address of the called terminal; the E.164 address of the called party is provided in the message.
- The gatekeeper sends back a Location Confirm message containing the required transport address.
- The gateway asks for permission to set up the call by sending an Admission Request to the gatekeeper. Upon reception of the Admission Confirm message, the gateway is ready to start the Q.931 setup phase.
- The gateway sends a Setup message on the signaling channel of the destination IP terminal.
- If the terminal is alive, a Call Proceeding message is sent back.
- The terminal asks for permission to set up the communication.
- The terminal sends an Alert message to the gateway indicating that the called user is being alerted of the incoming call. This may correspond to the usual ringing signal.
- The Alert message is forwarded to the ISDN part.
- The terminal sends a Connect message to the gateway indicating that the call is accepted. The Connect message contains the transport address needed for establishment of the H.245 channel.
- The terminal and gateway initiate the H.245 dialog for capability exchange and logical channels establishment.
- After the media channels are activated between the terminal and the gateway, the latter sends a Connect message to the ISDN calling party indicating that the voice communication can start.

It should be noted that this diagram depicts a typical scenario, but there exist shorter scenarios that use the Fast Connect procedure [1].

Media Control Functions

Once the connection is set up, the media control channel is used to carry all control information generated by the user or terminal. For voice communications, the main user-level control information is the DTMF tones used, for example, to interact with a voice server. Carrying these signals over a hybrid connection requires particular attention. The standard compression techniques used today for low bit rates introduce enough distortion to corrupt DTMF analog tones, making the receiving end system unable to correctly decode the original signals. Therefore, they need to be separated from the audio signal at the sender (if it uses a classical terminal attached to an adaptor) or at the gateway, and conveyed separately to the receiver.

Two approaches have been recommended by the VoIP Forum for carrying DTMF information. The first is to carry it in-band via RTP using a dedicated payload format. This has the advantage that the tones remain temporally synchronized with the speech. However, packet delivery is not guaranteed because of the unreliable transport protocol (UDP). Although packet loss can be kept very low in well-engineered networks and has negligible impact on voice quality, the loss of a DTMF tone can result in severe service malfunctions at the user level. The second solution uses out-of-band transport of DTMF signals on a separate and reliable media control channel. The drawback of this approach is that the signals loose their exact temporal position in the voice stream. This latter approach has been recommended for H.323 systems (i.e., using the H.245 channel). The protocol stack for the media control IWF is given in Fig. 7, that for the user plane IWF in Fig. 8.

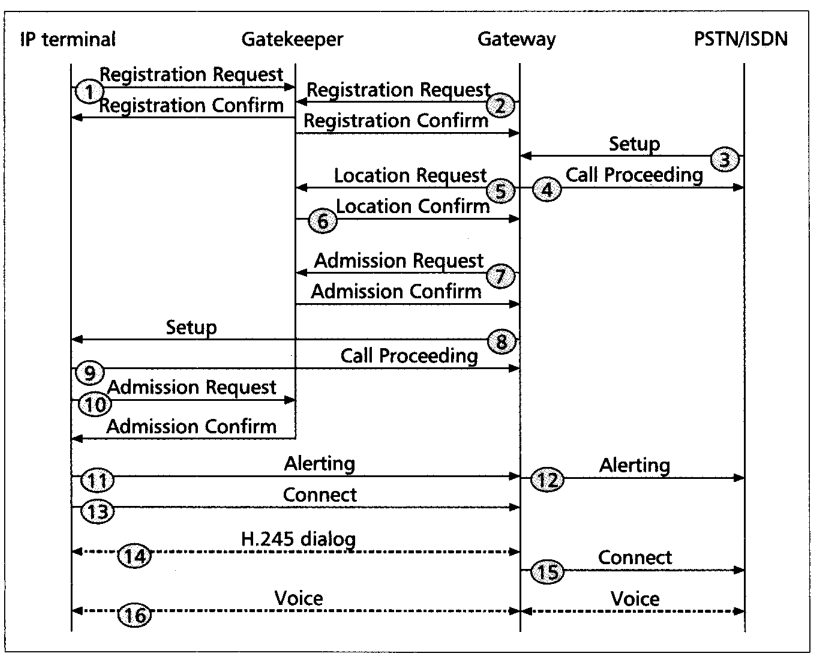
Media Adaptation Functions

The major user plane issue is to maintain the QoS required for voice connections. Instead of worrying about the quality of the transmitted bits, we focus on that of information delivered to the end user. Two main factors may influence the QoS experienced by the end user:

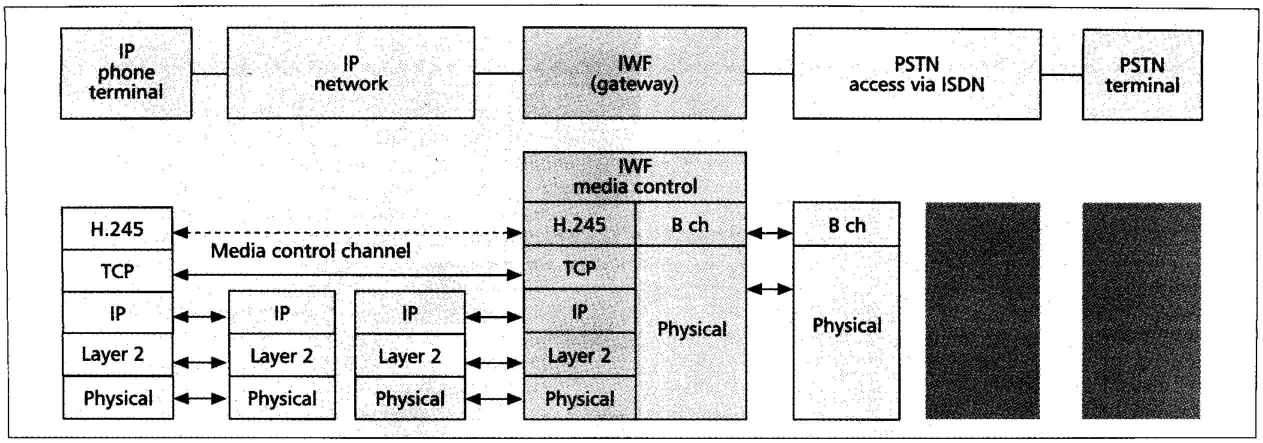
- The *end-to-end speech quality*, which may be affected by both the successive encoding/transcoding operations and the packet loss due to network congestion.
- The *end-to-end delay*, which mainly impacts the interaction between the participants of a conversation. It results from the coding/decoding process, packetization, and queuing delays.

On the IP network side, the service provider tries to accommodate a maximum number of voice connections at a time. Therefore, a key question arises: what are the appropriate mechanisms to be employed within both the end systems (including the gateways) and the network in order to optimize network utilization while maintaining the desired QoS for the end users?

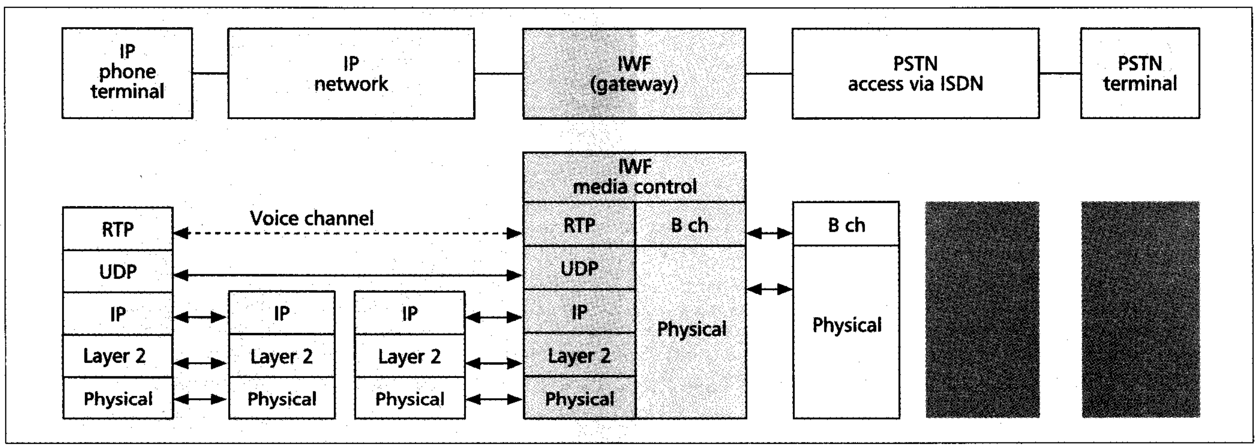
In this section, we first give a brief description of the techniques being implemented in IP networks in order to provide a certain service guarantee. Then we analyze how the end systems may influence the user-oriented QoS. We focus on



■ **Figure 6.** *Example of call control in a hybrid voice communication.*



■ **Figure 7.** *The IWF in the control plane: carrying DTMF signals.*



■ **Figure 8.** *The IWF in the user plane.*

the trade-off among bandwidth, delay, and computational complexity.

QoS and IP-Based Networks

While the PSTN network ensures a fixed delay and no-loss guaranteed service, this is not necessarily the case for IP-based networks. Indeed, services currently experienced on the Internet are best-effort services. They are characterized by the absence of any QoS specification at all. However, IP telephony applications will definitely need some kind of quality guarantees in terms of absolute delay, delay jitter, and packet loss.

The Integrated Services (*IntServ*) architecture was designed to provide a set of extensions to the best-effort traffic delivery model. For this purpose, it defines two classes: guaranteed service and controlled-load service [13].

- Guaranteed service (GS) provides a lossless transfer with tight delay bounds for flows that conform to the parameters negotiated at connection setup.
- Controlled-load service (CLS) yields a quality corresponding to a lightly loaded IP network at best effort; it is not expressed quantitatively. Admission control is based on the peak rate declared by a session initiator and on measurements of the load in the network. This could lead to higher network efficiency compared to admission control based only on declared source descriptors.

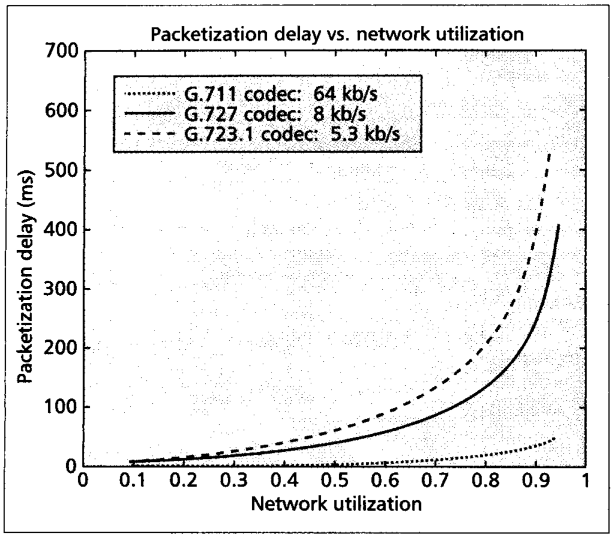
Both GS and CLS connections can be established by Resource Reservation Protocol (RSVP) signaling [14]. However, RSVP has some weaknesses that considerably undermine its wide deployment, mainly the soft-state reservation paradigm and the exponential growth of the reservation state tables.

In order to get around the weaknesses of the solutions proposed by the IntServ group, a new group, the Differentiated Services (*DiffServ*) group, was formed. They suggested that instead of maintaining the state of each and every flow, why not discriminate the packets according to their precedence? The precedence of a packet is indicated by the three first bits of the IP Type-of-Service field. This idea led to the concept of differentiated services, which also has the advantage of being "easily" implementable in existing networks.

As previously mentioned, the objective of a service provider is to increase network efficiency (reduce service cost) by accommodating as many voice connections as possible. This leads to higher packet loss ratios and delays. The sensitivity of IP voice services to data loss strongly depends on the mechanisms implemented in the end systems.

QoS and End Systems

The heterogeneity of networks causes voice traffic to be handled differently. Indeed, in the PSTN voice connections generally operate at the standard rate of 64 kb/s (pulse code modulated, PCM, signal or G.711). However, there is no need to keep such a high-bandwidth connection within the IP network. Rates ranging from 5.3 kb/s (G.723.1) to 8 kb/s (G.729) are usually more appropriate. The transcoding (PSTN to IP network) process occurs in the gateways. However, a lower bit rate will generally involve lower signal quality and higher delays. Indeed, while both the G.729 and G.711 coding stan-



■ **Figure 9.** *Packetization delay vs. network utilization.*

dards provide voice quality comparable to the usual telephone service quality (*toll quality*), an encoder based on the G.723.1 standard outputs a quality lower than toll quality. The introduced delay results from both a higher *processing delay* and an increasing *packetization delay*. The processing delay is the delay required to run the encoding algorithm on the uncompressed voice signal and create a stream of bytes ready to be sent to the packetization layer. The packetization delay represents the time needed to form a packet of compressed voice information of a given size. Therefore, when decreasing the encoding bit rate, the service provider can accommodate more voice connections at the expense of increasing signal distortion and delay.

As stated earlier, the RTP/UDP/IP protocol stack is used for the delivery of delay- and loss-sensitive services over packet networks. In such a scenario, each packet contains 40 bytes of pure header information (assuming no header compression technique is used). There is thus an inherent trade-off between packetization delay and payload-to-header ratio (channel utilization): the higher the payload-to-header ratio, the higher the packetization delay for a given encoding standard. For example, if the G.723.1 standard is used, 60 ms are necessary to collect 40 bytes of voice information (this corresponds to 50 percent channel utilization). Figure 9 illustrates the evolution of packetization delay versus network utilization. However, packetization delay can be dramatically reduced by multiplexing several voice connections in the same IP packet. A recent Internet draft proposes to perform this multiplexing at the RTP layer (e.g., appropriate for gateway-to-gateway communication in scenario 4).

The combination of processing, packetization, and queuing delays forms the *end-to-end delay* perceived by the end user. Increasing end-to-end delay may lead to better service implementation from the service provider viewpoint. However, this end-to-end delay, if strictly lower than 400 ms, should not affect the interaction between the participants of a conversation. Delays up to 150 ms require echo control, but do not compromise the effective interaction between the users.

Equivalently, the distortion introduced by both the successive encoding/transcoding processes and the data loss due to network congestion affect *end-to-end speech quality*. This quality must be equal or close to toll quality. Mechanisms such as error correction and error masking should be used in order to tolerate higher data loss while providing the same service quality. For example, G.723.1 interpolates a lost portion of the voice signal by simulating the vocal characteristics of the previous portion and slowly damping the signal [15]. The efficiency of an error masking scheme decreases when the number of packets lost at a time increases. Also, forward error correction (FEC) schemes have been proposed to alleviate loss bursts of a small number of packets. An RTP payload type for streams with FEC is being defined by the IETF. It should be noted that FEC introduces some predictable delay.

Although the relationship between all the factors influencing service quality and network efficiency is intrinsically complex, it is the key to implement an optimal voice service over IP networks.

Concluding Remarks

In this article we give an overview of the different technical issues involved in the provision of voice services over hybrid PSTN and IP networks. Due to the various aspects mentioned below, it is quite difficult to predict the pace at which this new technology will be accepted. There are a number of problems that still need to be solved; we discuss them briefly.

Complexity — The provision of voice services over the conventional PSTN is already extremely complex, and has led to highly sophisticated switches running programs of millions of source code lines. Five communication scenarios have been discussed in this article. This means that all of a sudden, the combinatorial complexity of voice services could be increased by a factor of five. It would be too optimistic to believe that this problem is solved by the intelligence-at-the-edge paradigm of the Internet. Indeed, as we have seen, many functions have to be implemented in the gateways and gatekeepers, which are centralized devices of the network. This means that even if the communications are established (and billed) properly for the five basic scenarios, it does not prove that this will be the case in more complex configurations; for example, what happens if a user wants to establish a call from a conventional phone to an IP terminal, which happens to have its calls forwarded to a cellular phone? If this works, will this call be properly charged?

Quality of Service — As we have seen, in a hybrid call the user data must go through a number of transcoding operations. It has not been proven that the user-perceived quality of service will be acceptable in a widely deployed hybrid service. Moreover, if at least one of the terminals happens to be mobile, the combination of the wireless problems with packetization delays and degradations due to transcoding can be quite challenging.

Ease of Use — It is clear that users appreciate the ease of use of universal communication systems such as the telephone and electronic mail, notably because of the simplicity of the addressing principle. However, in the case of hybrid voice services this simplicity will disappear, due mainly to the fact that end users will in some way be aware of the existence of intermediate devices (gateway, gatekeeper). As we have seen, there are proposals to make users oblivious to these issues by mechanisms based on aliases; however, these mechanisms can be highly vulnerable (e.g., to changes of telephone or Internet service providers).

Acknowledgments

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