Telephony over packet networks

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Transmission of voice and data over the same network brings more profitability and productivity to business communications. This long-time promised marriage of voice and data has now become a reality. In this paper, a brief description of the technologies of voice transport over packet-based data networks is presented. Furthermore, greater details are provided on IP telephony as the winning technology that promises to transform the communications world. This paper provides a global view of the technologies, norms and standardization efforts. It also touches upon the technical challenges faced by IP telephony in integration with SS7 and legacy networks (ATM, PSTN, PLMN), as well as quality of service issues.

L'acheminement de la voix et des données sur un seul réseau apporte à l'entreprise une rentabilité et une productivité accrues, un service rapide et efficace auprès des clients. Ce mariage longtemps promis de la voix et des données est devenu une réalité. Dans cet article on présente succinctement les différentes technologies de transport de la voix sur les réseaux de données et on détaille la plus prometteuse à savoir la téléphonie IP qui est une industrie émergente promettant la transformation du monde des communications. Cet article en donne une vision globale sur les technologies, les normes et les standards ainsi qu'un aperçu des organisations qui œuvrent dans ce développement. Il relate aussi les défis qu'il reste à résoudre pour la téléphonie IP comme l'intégration avec SS7 et les réseaux existants (ATM, réseaux fixes et mobiles), et la qualité de service.

1. Introduction

The Plain Old Telephone Service (POTS) is based on circuit switching. The major weakness is that the Pulse Code Modulation (PCM) voice is carried and switched on dedicated channels: during a call the established circuit is exclusively reserved to a caller. POTS limitations may be overcome by the packet telephony that can be defined as the voice transport over packet-based data networks. Since data traffic volume is increasing by leaps and bounds, voice and data integration represents one of many promising areas of development that brings improved profitability and productivity to business communications. It streamlines equipment and optimizes bandwidth.

2. Packet Voice Technologies

Packet voice technologies aim at the simultaneous delivery of voice and data over a unified data network. This convergence of voice and data is now a reality with three main solutions:

- Voice Over Frame Relay (VoFR),
- Voice Over ATM or VTOA (Voice and Telephony Over ATM) as referred by the ATM forum,
- Voice Over IP (VoIP).

The main factors that have contributed to this convergence are the availability of faster and cheaper digital signal processors (DSPs), and significant advances in standardized methods of compressing voice (Table 1). A voice call through the public switched telephone network uses 64 kbps of bandwidth, while the same call through a data network requires only 8 kbps or less with comparable quality (see table below). Also, the suppression of periods of silence improves voice transmission. The periods of silence can be up to 60% of the conversation.

A. Voice Over Frame Relay

Frame Relay is a standard communication protocol that provides the minimal set of features to deliver frames over a network of permanent virtual circuits (PVC). It is qualified as X.25 light since it is without error correction, and guarantees delivery in order to have optimized performance in terms of commutation and latency. This technology has many advantages:

- It is economical: the cost is roughly half that of X.25. In addition, the network coverage does not have any incidence on the invoicing; a wide area network is no more expensive than a local network.
- It is scalable: implementations are available for low bandwidths (e.g. 56 kbps), T1 (1.544 Mbps), T3 (45 Mbps) and even OC3-speed (155 Mbps).

The Frame Relay Forum has adopted many important standards. The FRF.11 known as Voice over Frame Relay Implementation Agreement, provides for bandwidth-efficient networking of

Table 1: Compression Methods

Type	Compression Model	Rate (kbps)	Algorithmic Delay (ms)	Comments
G.711	Pulse code modulation (PCM) of voice frequencies	64, 56 or 48	0	Uncompressed 64 kbps channel
G.722	SB-ADPCM (Sub-Band Adaptive Differential Pulse Code Modulation)	64, 56 or 48	0.125	
G.723	ADPCM (Adaptive Delta Pulse Code Modulation)	24 or 40	0.125	
G.726	ADPCM (Adaptive Delta Pulse Code Modulation)	16, 24, 32 or 40	0.125	
G.728	LD-CELP (Low-Delay Code-Excited Linear Prediction)	16	2.5	
G.729	CS-ACELP (Conjugate-Structure Code-Excited Linear-Prediction)	8	10	Quality similar to 32 kbps ADPCM. Default for VoFR.
G.729A	CS-ACELP (Conjugate-Structure Algebraic-Code-Excited Linear-Prediction)	8	15	Reduced complexity version of the G.729 codec.
G.723.1	Multi Rate Coder	5.3 or 6.3	37.5	MP-MLQ for the high rate coder ACELP for the low rate coder Default for VoIP.

Voice and Group 3 fax communications over Frame Relay, as well as defines multiplexed Virtual Connections. Two voice classes are supported.

- Class 1 calls use G.727 EADPCM at 32 kbps.
- Class 2 calls use G.729/G.729A CS-ACELP at 8 kbps.

The FRF.12 known as Fragmentation Implementation Agreement outlines how to break Frame Relay frames into smaller frames. By reducing latency, this agreement gives Frame Relay ATM-like capabilities, such as quality of service levels. The FRF.5 and FRF.8 standards tackle the service and network internet working with ATM.

B. Voice Over ATM

ATM (Asynchronous Transfer Mode) is the switching and multiplexing technology for supporting BISDN (Broadband Integrated Services Digital Network) services. It is the only standards-based technology that has been designed from the beginning as a layered architecture allowing multiple services like voice, data and video to be mixed over the network. This architecture is based on a three-layer protocol reference model:

- The physical layer transports the bits that make up ATM cells and converts them to appropriate electrical or optical format. The SONET (Synchronous Optical Network) technology is considered as the preferred physical layer.
- The ATM layer concerns the OSI level 2 (link). The transmission is based on a 53 bytes cell switching technology with 5 bytes for the header, and 48 bytes for the payload.
- The ATM Adaptation layer (AAL) assures the appropriate service characteristics and divides all types of data into the 48 bytes payload that will make up the ATM cell.

The following table (Table 2) provides a service classification and associated traffic types with ATM adaptation layer:

Table 2: ATM Services and Traffic Types

	Class A	Class B	Class C	Class D
Timing relation between source and destination	Required		Not Required	
Bit rate	Constant	Variable		
Connection mode	Connection oriented			Connectionless
AAL types	1	2	3/4, 5	3/4, 5

The ATM QoS Model is based on the following classes:

1. *Class A (or CBR) is a reserved bandwidth service for applications such as voice and circuit emulation. AAL1 is used in conjunction with Class A.*
2. *Class B traffic is defined as real-time Variable Bit Rate (VBR), has a bursty traffic where delay is critical, such as packetised phone calls. AAL2 is used in conjunction with Class B.*
3. *Class C traffic is non-real time VBR traffic, where delay is not critical, such as video playback.*
4. *Class D traffic is reserved for bursty and data traffic. It is composed of two classes: UBR (Unspecified Bit Rate) service similar to "best effort service". ABR (Available Bit Rate) service uses the available bandwidth.*
5. *AAL3/4 and AAL5 are used in conjunction with Classes C and D.*

The methods of the transmission of the voice over ATM are:

- AAL1: Circuit Emulation Service (CES)
- AAL2: Voice over ATM with VBR
- Proprietary solutions

The CES-IS (Circuit Emulation Service Interoperability Specification) defined in the recommendation af-vtoa-0078.000, covers the following types of CBR service:

1. Structured DS1/E1 Nx64 kbit/s (Fractional DS1/E1) Service
2. Unstructured DS1/E1 (1.544 Mbit/s, 2.048 Mbit/s) Service
3. Unstructured DS3/E3 (44.736 Mbit/s, 34.368 Mbit/s) Service
4. Structured J2 Nx64 kbit/s (Fractional J2) Service
5. Unstructured J2 (6.312 Mbit/s) Service

The problem with CBR service is that a virtual circuit is reserved whether voice is sent or not.

AAL2 allows defining more efficient voice service over ATM (class VBR) by supporting optimization techniques concerning voice compression and silence removal. The Voice and Telephony Over ATM to the desktop specification provides voice services to a broadband ATM terminal. The traffic is only 64 kbps PCM-encoded voice (without compression and silence removal). Each call uses one virtual channel.

The standardization effort is steered by the ATM forum. It has produced the specifications as shown in Table 3.

Table 3: ATM Forum - VTOA Specifications

Reference	Content
af-vtoa-0078.000	Circuit Emulation Service Interoperability Specification V2
af-vtoa-0083.000	Voice and Telephony Over ATM to the Desktop Specification
af-vtoa-0085.000	Specifications of (DBCES) Dynamic Bandwidth Utilization - In 64 kbit/s Timeslot Trunking Over ATM - Using CES
af-vtoa-0089.000	Voice and Telephony Over ATM - ATM Trunking using AAL1 for Narrowband Services Version 1
af-vtoa-0078.000	Circuit Emulation Service Interoperability Specification V2
ITU-T I.363.2	Voice Trunking on ATM including compressed speech support

C. Voice Over IP

Internet Protocol (IP) telephony is one of the hottest technologies in the telecommunications world. It is a rapidly growing industry that promises to transform the global communications area into a trillion-dollar market. A study from Killen & Associates (IP Telephony: new markets for systems and service providers) forecasts a $17 billion global market in the year 2002 for IP telephony equipment, software and services. The reason for this increasing interest is proven by the many advantages brought about by IP telephony's "packet voice" technology:

- Although most VoIP equipment today employs proprietary protocols, many vendors are beginning to support the ITU-T's Recommendation H.323 standard.
- The cost-quality trade-off can be interesting for users with urgent needs for simultaneous access to voice and data on remote connections.
- The coupling of VoIP with Advanced Intelligent Network (AIN) offers a significant innovation in the public network.
- Easy integration with World Wide Web technologies is enabling web-centric telephony service such as the "Call Me" button.

However many limitations exist:

- The architectural vision of fully integrated IP telephony networks within the business enterprise is not yet mature. Two concurrent

architectures have emerged:
- ITU architecture with H.323 protocols
- IETF architecture (SAP for announcements, SIP for signaling, RTSP for media-on-demand, SDP for describing media, etc.)

- Signaling interoperability problems: the signaling in the SCN is SS7, and in the IP network for example, the ITU standard H.323 or IETF's SIP (Session Initiation Protocol) is employed.
- Quality-of-service specific to large-scale telephony services: The public Internet has an unpredictable quality of service resulting in a negative impact on perceived quality. Typical multi-hop transmission delays can be as high as 500 ms or more.
- PC termination devices have delay limitation due to the continuing inefficiency of Windows 9x and NT.

H.323 protocols are the ITU multimedia communication standard for packet-switched networks that do not provide a guaranteed quality of service such as the Intranets and the Internet. The initial version of H.323 was designed to address IP voice within the enterprise.

Version 2 of H.323 protocols focus more on IP telephony in the wide area network (WAN). It includes faster call set-up and a new additional recommendation (H.450):

- H.450.1 - Generic functional protocol for the support of supplementary services in H.323
- H.450.2 - Call transfer supplementary service for H.323
- H.450.3 - Call diversion supplementary service for H.323

Version 3 of H.323 protocols addresses more issues. It includes more efficient call set-up by reducing message exchange, and has more supplementary services:

- H.450.4 - Call Hold
- H.450.5 - Call Park and Call Pick-up
- H.450.6 - Call Waiting
- H.450.7 - Message Waiting
- H.450.8 - Identification Services
- H.450.9 - Call Completion on Busy

Table 4: VoIP Standardization Groups

Organizations	Working Groups	Mission	Web site and Mailing List
IETF (Internet Engineering Task Force)	**MMUSIC** (Multiparty MUltimedia SessIon Control)	Development of Internet standards that support teleconferencing sessions (SAP, SDP, RTSP, SIP, RTSP, SCCP).	http://www.ietf.org/html.charters/mmusic-charter.html confctrl@isi.edu
	PINT (Public Switched Telephone Network and Internet Interfaces)	Definition of an architecture and protocols to support PSTN/Internet interworking.	http://www.ietf.org/html.charters/pint-charter.html pint@lists.research.bell-labs.com
	IPTEL (IP Telephony)	Definition of a voice call processing syntax (CPL) and a gateway location protocol (GLP).	http://www.ietf.org/html.charters/iptel-charter.html iptel@lists.research.bell-labs.com
	Megaco (Media Gateway Control)	Definition of the architecture and requirements for controlling Media Gateways from external control elements.	http://www.ietf.org/html.charters/megaco-charter.html megaco@baynetworks.com
	Sigtrans (Signaling transport)	Definition of a "Signaling Transport" protocol over IP networks. Signalling can be Q.931 or SS7 ISUP.	http://www.ietf.org/html.charters/sigtran-charter.html sigtran@baynetworks.com
ITU (International Telecommunication Union)	**Study Group 16** (Multimedia services and systems)	Q.13/16: Packet-switched multimedia systems and terminals	http://www.itu.int
	Study Group 16 (Multimedia services and systems)	Q.14/16: Common protocols, MCUs and protocols for interworking with H.300-series terminals	http://www.itu.int
	Study Group 15 (Transport networks, systems and equipment)	(Q.21/15) Transport Network Equipment for Interconnecting GSTN (General Switched Telephone Network) and IP Networks	http://www.itu.int
ETSI (European Telecommunications Standards Institute)	**TIPHON** (Telecommunications and Internet Protocol Harmonization Over Networks)	IP and PSTN/ISDN/GSM interoperability. TIPHON specifications are tested over a verification network called TIPHON net.	http://www.etsi.org/tiphon and http://docbox.etsi.org/tech-org/tiphon/Document/tiphon
ANSI (American National Standards Institute)	**TIA TR-41.3.4**	Performance and Interoperability Requirements for Voice-over-IP Telephone Terminals	www.tiaonline.org
IMTC (International Multimedia Teleconferencing Consortium)	**The VoIP forum**	Support of the development and implementation of interoperable multimedia teleconferencing solutions based on open international standards.	www.imtc.org
		Development of the VoIP interoperability agreement.	voip@imtc.org
ITC (Internet Telephony Consortium)		Support of the research about technical, economic, strategic and policy issues that arise from the convergence of telecommunications and the Internet It is not a standard forum like VoIP, IETF, IMTC and ETSI.	http://itel.mit.edu/
VON (Voice on the Net)		The VoN is a trade group that organizes trade shows and conferences about Internet telephony technologies.	www.von.org

Although H.323 is a worldwide standard signaling protocol, there is another alternative to handle signaling for IP telephony. It is based on the proposed standard IETF'Session Initiation Protocol (SIP) which can be used, with its extensions, to address the services of intelligent network as described by the ITU AIN recommendations (Q.1211 and Q.1221). Its major strength is it is a lightweight protocol that better fits in the philosophy of Internet protocols.

To inter-operate with the circuit-switched network, an IP telephony gateway must support the Signaling System 7 (SS7) protocol. SS7 is used for basic call setup, management, and tear down, and to query databases which support Intelligent Network services. The support of SS7 by IP telephony is an important step in the integration of the SCN and IP data networks. To do this, several efforts have sprung up assuming a distribution of call control:

The IPDC (Internet Protocol Device Control) is a series of protocols for signaling transport within an IP network, device management, media control and connection control.

The SGCP (Simple Gateway Control Protocol) is designed for controlling Voice over IP (VoIP) Gateways from external call control elements.

The MGCP (Media Gateway Control Protocol) is resulted from the fusion of the SGCP and IPDC proposals.

Diameter is a lightweight policy protocol used between a client and a server for authentication, authorization and accounting of various services, which is integrated in the IPDC proposal.

The OSP (Open Settlement Protocol) constitutes a set of protocols that aims at inter-domain authentication, authorization and accounting for IP telephony enabling ISPs to scale their VoIP services.

The GLP (Gateway Location Protocol) is being developed to support discovery and location of gateways in remote administrative domains.

Other protocols are being discussed in the IETF as SigTrans for signaling transport across IP and Megaco for device control.

The standardization effort is steered by many organizations. We describe them briefly in Table 4.

Quality of Service (QoS) is an important issue, especially for time-sensitive applications. The over-provisioning of the bandwidth can be a solution for the QoS scarcity in IP networks. Although there are new technologies such as the dense wave division multiplexing (DWDM) and Gigabit Ethernet that can provide the increase in bandwidth, analysts agree that the QoS mechanisms are needed.

The three main architectures and techniques defined for IP QoS are:

- The Integrated Services in the Internet: **intserv**
- The Differentiated Services: **diffserv**
- The Multiprotocol Label Switching: **mpls**

The Integrated Services architecture for the Internet (intserv) was proposed in RFC 1633 in order to support real-time traffic as well as "best-effort" traffic. It is relatively complex. It requires explicit resource reservation, admission control, packet classification, and scheduling and has difficulties in scaling to large backbones. This architecture is based on the Resource Reservation Protocol (RSVP).

ATM can support intserv. ISSLL, a group in the IETF, looks at how intserv can map to specific link layers. It has a sub-group that is looking at the mapping of intserv to ATM.

The differentiated services (diffserv) model minimizes signaling and concentrates on aggregated flows. It is based on a new implementation of the IPv4 type of service (ToS) header field or the IPv6 Traffic Class octet, renamed Differentiated Services (DS), and used to mark a packet to receive a particular forwarding treatment, or per-hop behavior, at each network node. There is no current work on mapping diffserv to work with ATM transport.

Recently, the diffserv group has adopted a definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers (RFC 2474) and architecture for Differentiated Services (RFC 2475).

The label-based switching aims to set up virtual channels for traffic that does not need to be routed at each hop. Implementations can be based on flow detection (for example, IP switching) or routing topology (for example, Tag switching). The most important feature of this approach is the mpls called tag switching. Mpls labels or tags each packet to define a class of service. This means that there is no need for further analysis on the header. In the context of ATM networks, a virtual path and a virtual channel (VP, VC) characterize a tag. Mpls offers an integration of the IP protocol routing scheme with the cell switching technology, simplifies IP and ATM interoperability, and improves performance, scalability, and functionality. The mpls group has not yet produced any Request for Comments.

3. Discussion

We have reviewed the current packet voice technologies in this paper (see Table 5 for a comparison). Corporations can evolve their networks from separate costly networks carrying voice and data traffic to consolidate packet infrastructure, increasing flexibility and productivity.

Table 5: Packet voice technologies comparison

Technologies	Strengths	Weaknesses
Frame Relay	* Mature technology * Wide area networking (WAN) technology used for multisite connectivity * Savings of 40% to 60% in comparison with comparable leased lines and X 25 services * Interwork with ATM * Simple to deploy * Low overhead * Inexpensive * Short investment return time (usually less than a year)	* Basic QoS * Pre-configured connections * Less popular out of North America
ATM	* Better backbone technology for both wireless and wireline voice traffic * The best multiservice networking technology * Large services categories ranging from best effort (UBR) to reserved bandwidth (CBR) * Highly scalable (from 1 5 Mbps to 1 2 Gbps and more) * Flexible bandwidth management	* High overhead (cell tax) and inefficient cell padding * Complex * Expensive * Difficult to find ATM expertise * Interoperability problems between vendors
IP	* Broadly deployed * De-facto desktop standard * Meshed connectivity * Availability of the expertise	* High overhead. * Lack of adequate QoS management, accounting, and control or vendor specific * Standards not yet completely finalized

ATM is the underlying technology for high speed networks. It combines the reliability of circuit switching with the efficiency of packet switching.

Depending on the case, ATM can be considered as a full enabling technology for voice with VTOA, or as a layer 2 for IP network but the second scheme needs an adequate mapping between IP QoS and ATM QoS. ATM is not ready for widespread use at desktop from a market point of view, but ATM can be used as a transport technology to make

IP networks reliable. The coupling of IP routing and switching is promising. The VTOA solution is adequate only for enterprises that have full ATM LAN with ATM desktops.

As regards Frame Relay, it is positioned as an access service to ATM and intra-company networking technology. Although VoFR has reached maturity, it is being overtaken by VoIP which targets both residential and corporate markets.

Despite the technological challenges, VoIP is receiving an intensive and growing attention from the industry and the users. New generation Telcos has sprung up like I-Link, Level 3 and IDT(Net2Phone). The main reason of this success is the popularity of IP, its domination at the desktop with lower costs and its transport independence feature (Ethernet, PPP, Frame Relay, ATM, SONET, etc.). However, there are architectural issues for IP telephony. The Internet Engineering Task Force (IETF) is defining an open architecture keeping intelligence out of the endpoints and facilitating deployment, billing and metering requirements. Also, the following issues still need to be solved:

- Interoperability with legacy networks and among all the Voice over Packet Protocols (matching VoFR or VoATM virtual channel to the VoIP session, etc.).
- Reliable and efficient signaling transport over IP (ISUP/TCAP over IP, etc.).
- Quality of Service challenges for IP networks.
- Quality of Service routing.
- Address resolution: SIP URL, H.323 URL, E.164 address.
- VoIP in the wireless environment.
- Pricing model for the IP Telephony based on QoS offered.
- Economic and regulatory issues.

6. Glossary of terms used

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
AIN	Advanced Intelligent Network
ACELP	Algebraic-Code-Excited Linear Prediction
ADPCM	Adaptive Delta Pulse Code Modulation
ATM	Asynchronous Transfer Mode
BISDN	Broadband Integrated Services Digital Network
CBR	Constant Bit Rate
CELP	Code excited linear prediction
CES	Circuit Emulation Service
CS-ACELP	Conjugate-Structure Algebraic-Code-Excited Linear Prediction
DSCP	DiffServ Code Point
DSP	Digital Signal Processing
ETSI	European Telecommunication Standards Institute
FRF	Frame Relay Forum
IPDC	Internet Protocol Device Control
ISSLL	Integrated Services over Specific Link Layers
GLP	Gateway Location Protocol
GSTN	General Switched Telephone Network
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSEC	Internet Protocol Security
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
QoS	Quality of Service
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
LD-CELP	Low-Delay Code-Excited Linear Prediction
LPC	Linear Predictive Coding
MIB	Management Information Base
MGCP	Media Gateway Control Protocol
MMUSIC	Multiparty Multimedia Session Control
MOS	Mean Opinion Score
MP-MLQ	Multipulse Maximum Likelihood Quantization
MPLS	Multiprotocol Label Switching
OC-n	Optical Carrier
POTS	Plain Old Telephone Service
PPP	Point to Point Protocol
OSI	Open Systems Interconnection
PCM	Pulse Code Modulation
PSTN	Public Switched Telephone Network
RAS	Registration Admission and Status
RPE-LTP	Regular Pulse Excitation Long-Term Predictor
RSVP	Resource Reservation Protocol
RTP	Real Time Protocol
RTSP	Real-Time Streaming Protocol
SAP	Session Announcement Protocol
SB-ADPCM	Sub-Band Adaptive Differential PCM
SCN	Switched Circuit Network (GSTN, ISDN
SDP	Session Description Protocol
SGCP	Simple Gateway Control Protocol
SIP	Session Invitation Protocol
SONET	Synchronous Optical Networks
SRP	Scalable Resource Reservation Protocol
SS7	Signaling System 7
TCP	Transmission Control Protocol
ToS	Type of Service
TLS	Transport Layer Security
UDP	User Datagram Protocol
URL	Uniform Resource Locator
VC	Virtual Circuit
VP	Virtual Path
VoFR	Voice Over Frame Relay
VoIP	Voice Over IP
VTOA	Voice and Telephony Over ATM

7. For further information

1. Daniel Minoli, Delivering Voice over IP Networks. John Wiley & Sons, April 1998;

This book addresses the engineering challenges of setting up an IP voice network.

2. Daniel Minoli and Emma Minoli, Delivering Voice over Frame Relay and ATM. Wiley Computer Publishing, May 1998;

This book examines how to move voice over Frame Relay and ATM networks, and offers an extensive case study exploring both benefits and drawbacks.

The Frame Relay Forum:

http://frforum.com: This web-site provides tutorials, presentations, white papers, market surveys and technical briefs for FR technology.

The ATM Forum:

http://www.atmforum.com: This web-site provides educational information and market surveys for ATM technology.

Internet Telephony Resources:

1. http://www.telephony.net: This non-commercial web site provides links to vendors, next generation telcos, standards, protocols development in the IP telephony world.

2. http://www.fokus.gmd.de/research/cc/glone/projects/ipt/: This website provides the state of art on the Internet telephony with links to standardization bodies, products, research contributions, tutorials.

About the author

Said Soulhi got an engineering diploma in 1982 from the National Polytechnic Engineering School of Electrotechnology, Electronics, Computer Science & Hydraulics of Toulouse (ENSEEIHT) and received a Ph.D. in Computer Science at Paul Sabatier University at Toulouse (UPS) in 1985. He acted as information technology consultant for the industry and as telecommunication network management software developer at ALCATEL TELECOM. He is currently a Senior Researcher at CRIM (Computer Research Institute of Montreal) and Associate Professor at École de Technologie Supérieure. He heads the telecommunication technologies group at CRIM. His interests are in the area of distributed systems architecture, Internet realtime and multimedia services and protocols, intelligent networks, modeling and analysis of computer-communication networks.