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AudioLAN : Internet Technology Inside Voice Communication Systems

Gilles Gawinowski* & Laurent Guichard**
Eurocontrol Experimental Centre
France

Abstract

With the initiation of studies in 1996, AudioLAN has been a pioneer project in voice-over-IP technology for Voice Communication Systems (VCS). The Eurocontrol Experimental Centre (EEC) was the first player of the Air Traffic Control (ATC) community to understand the potential of this technology. Deployment of a simulation environment for the EEC; Instilux Luxembourg; Paris Roissy CDG Airport; Ecole Nationale de la Navigation Aérienne (ENAC); French, Turkish, Hungarian and South African Civil Aviation Authorities; and the Military ATC, began in 1998. Today, AudioLAN is working within the operational environment, with the potential to support 470 positions and six license agreements with industrial organizations for commercialisation.

The ATC operational context is introduced and Voice-over-IP (VoIP) concepts are described. A technical description of the AudioLAN architecture and functions in both simulation and operational environments follows. The paper concludes with a brief description of future R&D, providing direction to manufacturers and users to focus on unique application-specific challenges and opportunities and " value-added " points of distinction for the emerging Internet Telephony market.

Keywords: Internet Telephony; Internet Protocol (IP); VoIP (Voice-over-IP); Voice Communication System; PSTN (Public Switched Telecommuni--cation Network); ISP (Internet Service Provider); GateWay; GateKeeper; H323; CTI (Computer Telephony Integration).

Introduction

Voice Communication Systems for ATC simulators are generally derived from 'operational' environments. That is, these systems originate in the telephone world and are primarily based on hardware components.

In the 'real' ATC world, VCS's are generally digital, centralised and based on a non-blocking PCM (Pulse Code Modulation). These PBX (Private Branch Exchange) systems operate much like mainframes, with all their attendant characteristics. They are proprietary and closed with very complex architectures, there is slow evolution of new features, and they are priced accordingly (even if they are based on high telecom technology, such as PCM, ISDN, and voice compression).

Due to the latest evolution of communication and information technologies through the Internet Technology wave, it is now possible to implement radio-telephone applications using the Internet Protocol (IP), pure software components and mass-market computers. The AudioLAN system, designed and developed by the EEC, is based on these new technologies (Gawinowski & Goguey, 1998).

The primary objectivess of the AudioLAN project were:

1) to replace the hardware-based PBX by network equipment and a software PBX : IPBX, and
2) to replace specific operator position hardware by Commercial-off-the-Shelf (COTS) software for PC's or workstations.

Since 1996, the Internet Telephony broker has dominated the network and telecom market, from telecom manufacturers to PSTN and ISP operators. The current challenge is to provide more value-added networking services within constrained budgets. Merging voice and data networks is a step toward meeting these requirements and VoIP is emerging as a key technology supporting this goal. The convergence of voice and data networks means that PSTN users will benefit from rapid innovations in Internet Protocol over the next ten years.

There has been a progression of these types of services and a migration toward IP solutions. In 1996, terminal (end-system) applications were commercialised, followed by Gateway applications in 1997 and, finally, GateKeeper applications in 1998. In 1999, a first generation global and integrated solution and a significant deployment of software-based PBX services is expected (Susbielle, 1998).

The advantages of such a solution are :

- Flexibility and evolution capability
- Voice and data convergence
- Reduced operating cost
- Unified network management
- Standards-based Internet Technology solutions
- Scalability to present and future needs
- PBX integration via CTI (Computer Telephony Integration)
- Call Centre or added-value services
- Browser-based implementation.

In this context, AudioLAN is a unique Voice Communication System in the ATC world in proposing and implementing the use of VoIP technology in the simulation environment. This innovative system may also be the solution of the future for the operational world.

IP Telephony concept

IP Telephony includes multiple features. In addition to the capability for establishing voice communication between two individuals, IP telephony also encompasses all the features of the classical switched telephony network plus features from Internet telephony (e.g., Phone-to-Phone, Fax-to-Fax, PC-to-Phone, Phone-to-PC, PC-to-Fax, Email-to-Fax, Fax-to-Email, and Email-to-

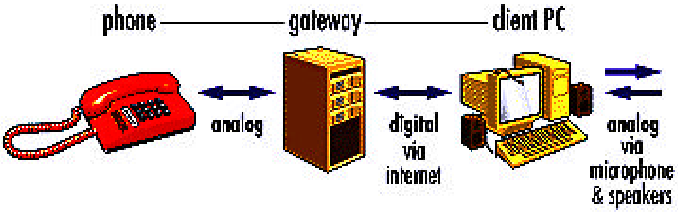

Figure 1. An IP-based voice communications concept.

Voice recognition). The primary goal is to unify the transport of information around IP. At present, IP telephony appears to be the only medium that can support convergence between telephony and computers. An overview of an IP-based telephony concept is given in Figure 1.

With the growth of the Internet and the deregulation of the telephony market, IP telephony appears promising as an alternative to the classical PSTN. At present, all Internet providers or telephony operators are developing IP telephony products and services. For example, Microsoft is developing Netmeeting, and CISCO and ALCATEL provide gateways between the IP and PSTN Networks for voice applications.

Using Existing Infrastructures

IP appears to be the clear protocol to use for computer communications ; using this existing technology provides many advantages. It is based on IP, the Internet Protocol that is available across all networks. In addition, because IP is a well-known technology and is used around the world, it has become very inexpensive to implement. IP telephony also does not requires a significant investment. That is, to deploy an IP telephony network, one need only purchase gateways that interconnect the new network with the classical PSTN. These gateways may also replace the classical PBX by providing a software emulation of its functionality.

IP Telephony protocols

Voice and data have opposing transmission requirements. Voice requires a constant stream and a controlled latency time, and loss of data is permitted. In contrast, data communication does not require a constant stream or controlled latency time, but does not allow data loss.

The primary issue with regard to voice transportation is a guaranteed constant bandwidth (Quality of Service [QoS] guarantee) for real-time transportation without distortion. Latency time (end-to-end transmission time) must be less than 100 ms. Today, IP is the universal protocol. But is IP able to meet voice transmission requirements ; that is, can IP support real-time and reliable voice transmission ?

Given the TCP/IP protocol is unable to guarantee real-time transport of data, it is, therefore, necessary to use the UDP/IP protocol. Two other protocols are used with UDP/IP : RTP (Real-Time Protocol) detects the loss of data and maintains the chronological order of the data, and RTCP (Real-Time Transport Control Protocol) provides session control. An implementation based on the IP, RTP and RTCP protocols is able to perform real-time transmission of voice on non-controlled media such as the Internet.

This implementation is even more effective when using bandwidth-controlled network technologies, such as ATM or switched Ethernet, in that it supports IP-embedded voice information trans-mission on any physical layer (e.g., Internet, Ethernet, ISDN, ATM).

Our implementation, AudioLAN, must be considered independent of the network layer with regard to the Quality of Service (e.g., Constant Rate, latency time) that it can assume. Quality is guaranteed from point-to-point through LAN and WAN.

Voice and Data Convergence

With VoIP, data and voice convergence is accomplished through CTI with IP as the point of convergence. CTI's are applications that provide integration between the computer and the telephone. With IP telephony, development of a CTI would be easier and would cost less, given that the computers would be able to manage both voice and data. (In an ATC environment, an example of CTI would be the integration of telephone/radio facilities in the controller display environment (Karsety & Pecaut, 1998).)

AudioLAN Architecture and Functions in the Simulation Environment

In the simulation environment, AudioLAN runs within a pure IP infrastructure. The AudioLAN terminal presently runs on a Sun Workstation under Unix Solaris using a Java-sensitive touch screen.

AudioLAN consists of three main components :

- A Human-Machine Interface (HMI),
- A Communication Manager, and
- A Voice Manager.

An overview of the Simulation Environment AudioLAN is given in Figure 2.

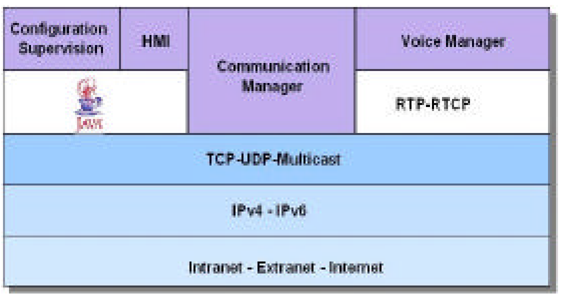

Figure 2. AudioLAN architecture within the Simulation Environment

Human-Machine Interface

A graphical HMI, based on Java, is provided by a dedicated touch input panel ; this may also be integrated with other HMI functions on a general purpose screen. The HMI layout is adaptable and is context-sensitive to the operational requirements.

Communication Manager

The Communication Manager component manages the following telephone and radio functions :

- Direct Call, Intercom, Conference, Transfer, Diversion, Group Hunting, and Multiple Addressing;
- Radio Transmit/Receive and Push-to-Talk;
- Radio coupling;
- Volume +/- and Buzzer;
- Radio/Telephone splitting on headsets and microphone;
- Silence Threshold Adjustment;
- Statistics; and
- Auto-test.

Voice Manager

The Voice Manager manages two simultaneous, independent full-duplex voice channels (one for radio and one for telephone). The following voice compression standards are available : PCM, PCM2, PCM4, IDVI, DVI2, DVI4, GSM and LPC4 (64 kbps to 9.6 kbps transfer rates).

Radio and telephone communications can be performed with separate headset (for radio) and handset (for telephone) or with a single headset and a radio-telephone three-mode splitting mechanism (radio only, telephone only, or radio-right/telephone-left modes).

Cost, Quality and Performance in the Simulation Environment

The cost for an AudioLAN terminal within the simulation environment is about 4K euros (~$4020) (including the workstation and touch-sensitive screen). CD-audio quality is provided and end-to-end latency time is about 60 ms, due to buffer and jitter mechanisms.

AudioLAN Architecture in the Operational world

In the real world, AudioLAN is running on an IP infrastructure on the LAN side and may be interconnected to the PSTN and radio for the WAN part. A graphical depiction of the AudioLAN architecture is given in Figure 3.

To interconnect the IP/PSTN world, AudioLAN is using an H323 Architecture composed of:

- A Terminal;
- A Gateway; and
- A GateKeeper.

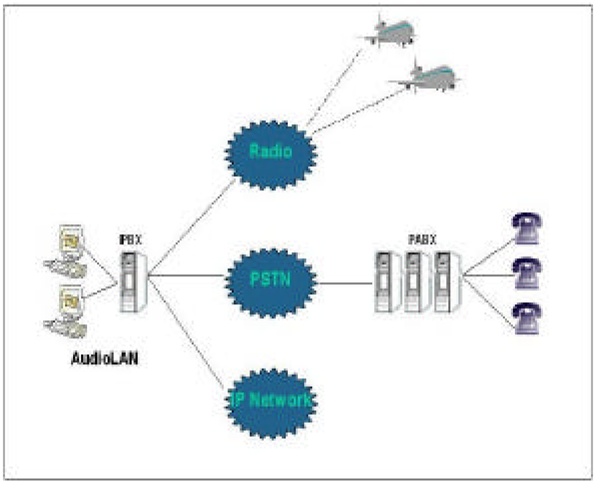

Figure 3. AudioLAN Architecture in the Operational World

Terminal

The Terminal is the Operator Position dedicated to the user to use radio and telephone facilities.

Gateway

Gateways provide the ability to match up dissimilar devices (for example, a plain voice-only telephone with a multimedia terminal). Within AudioLAN, the IP Telephony Gateway is the computer that provides the physical interface between PSTN (e.g., analogue, ISDN, etc.) and the IP (e.g., Ethernet, etc.).

GateKeeper

A GateKeeper provides address resolution (e.g., a phone number to an IP address) and also sets limits on bandwidth used.

The AudioLAN Gatekeeper provides the gateway's " intelligence ." An H323 separates hardware and software in the IP telephony architecture. The GateKeeper is the software companion of the Gateway and represents the functionalities of the software PBX.

The GateKeeper performs the following functions :

- Call control signalling ;
- Call management and domain control ; and
- Address translation.

The EEC has developed a basic GateKeeper to interconnect AudioLAN to the PSTN/radio 'world' and has built an IP telephony platform prototype to demonstrate IP Telephony within an IP/PSTN/radio infrastructure. GateKeeper development is based on the Natural Micro System

Gateway and the RadVision H323-stack.

Cost, Quality and Performance in the Operational Environment

The cost of a Gateway for a T1 line (i.e., 2 Mbps) is approximately 10K euros (~$1030). Typical tele-phone quality is provided and end-to-end latency time is approximately 120 ms, due to buffer and jitter mechanims and processing on the IP/PSTN Gateway. New DSP cards are expected to improve significantly the latency time.

Conclusion

AudioLAN has demonstrated that voice and data may be integrated and transmitted via network using Internet Protocols. It proves that IP telephony is not just a dream and that ATC can benefit from this new technology. The development of the software layer, also called GateKeeper, on the gateway is complex and its difficulty depends on the functionalities that would be implemented into the IP telephony network.

The first goal is to implement the basic functions of a PBX, keeping in mind that the solution must be as generic as possible. Then high-level functionalities could be developed to finally build a complete ATC IP telephony network.

The IP telephony market is young but it is expanding rapidly. Companies should invest in IP telephony to secure a role in the potentially large market and to propose new value-added services. Considering IP telephony in operational voice communication systems is no more imaginary than considering VHF radio over IP or perfect voice and data integration.

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* Mr. Gilles Gawinowski is the head of the Telecom & Technology Multimedia Center of Expertise at Eurocontrol's Experimental Center in Paris. He can be reached at Eurocontrol Experimental Center, Rue des Bordes, 91220 Bretigny/Orge Cedex, France. Email: Gilles.gawinowski@eurocontrol.fr, Phone: +(33) 1 69 88 74 46.

**Mr. Laurent Guichard is a project leader at Eurocontrol's Experimental Center in Paris. He can be reached at Eurocontrol Experimental Center, Rue des Bordes, 91220 Bretigny/Orge Cedex, France. Email: Laurent.guichard@eurocontrol.fr, Phone: +(33) 1 69 88 74 39.