Europe's air traffic control on the way to voice over IP

Voice communications systems will continue to be absolutely essential for the safe handling of air traffic in Europe in the future. But the rapid technical development opens up completely new opportunities for air traffic control (ATC) organizations. They are subjecting these new methods to thorough tests, for high safety requirements must be met. This careful migration is made possible by the radios of the R&S®Series4200 family, because they support several communications technologies that make migration easier.

Tried and tested in changing times

All flights based on instrument flight rules (IFR) require a constantly available voice connection between pilot and air traffic controller. For this reason, voice communications are one of the most important working bases in air traffic control. The technical basis for voice communications is a radio network that ensures countrywide coverage during all phases of the flight independent of altitude. For this purpose, radio stations with radios and associated RF components, such as transmitter/receiver filters, couplers and antennas, are installed at suitable radio locations, e.g. at airports or on mountaintops. Depending on the operational requirements, a number of radio channels are available for each location, combined with the corresponding number of device installations. To ensure the availability of the radio system in case a component fails, additional devices are installed to provide redundancy.

The radio systems as well as the air traffic controllers with their controller working positions (CWP) are connected with the terrestrial voice communications system (VCS). The VCS connects the controller working positions dynamically or semipermanently with the associated radio systems so that the air traffic controllers can work on the frequency assigned to their respective sectors. The assignment of a CWP to a

radio channel can change depending on the time of day and the traffic volume. From a technical and organizational perspective, it is also possible that sectors are taken over by air traffic controllers who are working at other locations.

The connection of the radio systems and controller working positions or the interconnection of individual voice switching nodes is made using analog telephone lines or via digital transmission lines with 2 Mbit/s in line with the ITU-T G.703 standard. The transmission lines for connecting radio locations are leased by the ATC organization from a telecommunications provider. However, many providers in Europe no longer offer analog landlines, since they can hardly be operated economically anymore. In the near future, the digital transmission technologies that have been used for years, such as plesiochronous digital hierarchy (PDH) and synchronous digital hierarchy (SDH), will also no longer be available for connecting remote radio stations to the voice switching system.

Alternatives are in demand – and also in sight: Voice over IP (VoIP) is an obvious choice, due to its widespread use and the many years of experience with this technology in the classic telecommunications environment. VoIP is not only a replacement for analog or PDH transmission technology; it also offers several significant advantages.

R&S®Series4200 software defined radios

The R&S®Series4200 is the latest generation of digital software defined radios for stationary use in civil and military air traffic control. The possible applications range from small emergency radio systems with only a few channels to nationwide radiocommunications systems with several hundred channels.

- VHF frequency range from 112 MHz to 156 MHz
- UHF frequency range from 225 MHz to 400 MHz
- 50 W transmit power in VHF and UHF range
- Automatic main/standby operation
- USB service interface for configuration and software downloads
- Remote control and remote monitoring via Ethernet interface
- Suitable for data transmission in VDL mode 2 standard
- Connection via E1 interface
- Voice over IP via software upgrade

VoIP: communications technology of the future for air traffic control

VoIP systems no longer transmit voice over a circuit-switched voice network, but over an IP-based packet-switched data network. For this purpose, the voice signal is digitized at the source and divided into IP data packets. The data network consists of routers that handle the routing of the voice packets and the data packets on the basis of the IP address. At the information sink, the digital voice packets are converted back into an analog signal.

Cost savings through integrated voice and data networks

ATC organizations already operate extensive data networks, e.g. for the transmission of radar and flight plan data. Therefore, it seems reasonable to transmit voice and data over a single network in the future, as is possible with VoIP. At remote locations, the integration of voice and data into a common network is particularly advantageous, for it also saves money – only a single network must be planned, installed and operated.

Data networks that should also transmit digital voice information must meet special requirements, for this information must be transmitted in realtime and with absolute reliability. The radios are remote-monitored and remote-controlled via an IP-based data connection. In the future, voice transmission as well as remote monitoring will run via the same IP interface on the radio.

Simple setup

While the technical planning of the data networks as described above is more complex, the installation of the VoIP network components simpler. This is because standard cabling and components, which are already in use for today's LAN technology, are sufficient. All security mechanisms known from the data world, such as Internet protocol security (IPSec), and prioritization techniques, such as differentiated services (DiffServ, RFC 2474, RFC 2475) or multiprotocol label switching (MPLS), can also be used for voice transmission via VoIP.

Shift of routing technology into the IP network

Another advantage of VoIP systems is the possibility of doing without central switching nodes – such as those required in circuit-switched voice networks. For security reasons, today's routing technology is designed redundantly in the voice network – which results in added costs. In an IP-based data network, the data packets are not distributed by a central unit but by routers whose number and performance are determined by the size and requirements of the network. The router decides, based on the IP address of the information sink, where the packet should go. Consequently, the information source only needs to know the IP address of the communications partner. The addresses defined in an address plan that can dynamically be adjusted to the operational conditions. Therefore, the data packets are switched via several routers distributed in the network, and not centrally. The advantage is that existing redundancy mechanisms in the IP network can be utilized.

Implementation of new functions

Speaking in favor of the introduction of VoIP is not only that it replaces transmission methods which are no longer available, but also that new functions can be implemented in the voice communications system – functions that are difficult to implement using today's circuit-switched technology.

One example are the functional airspace blocks (FABs) planned within the framework of the "Single European Sky" initiative, in which several countries are united into one FAB, e.g. the FABEC (FAB Europe Central), comprising France, Belgium, the Netherlands, Luxemburg, Switzerland and Germany. These FABs no longer organize the airspace over Europe according to national borders, but with a view toward achieving the highest possible efficiency in utilizing the airspace so that air traffic can be handled in a more time- and cost-saving way.

One of the many requirements for an FAB is the harmonization of the communications technology to ensure sufficient

VoIP infrastructure

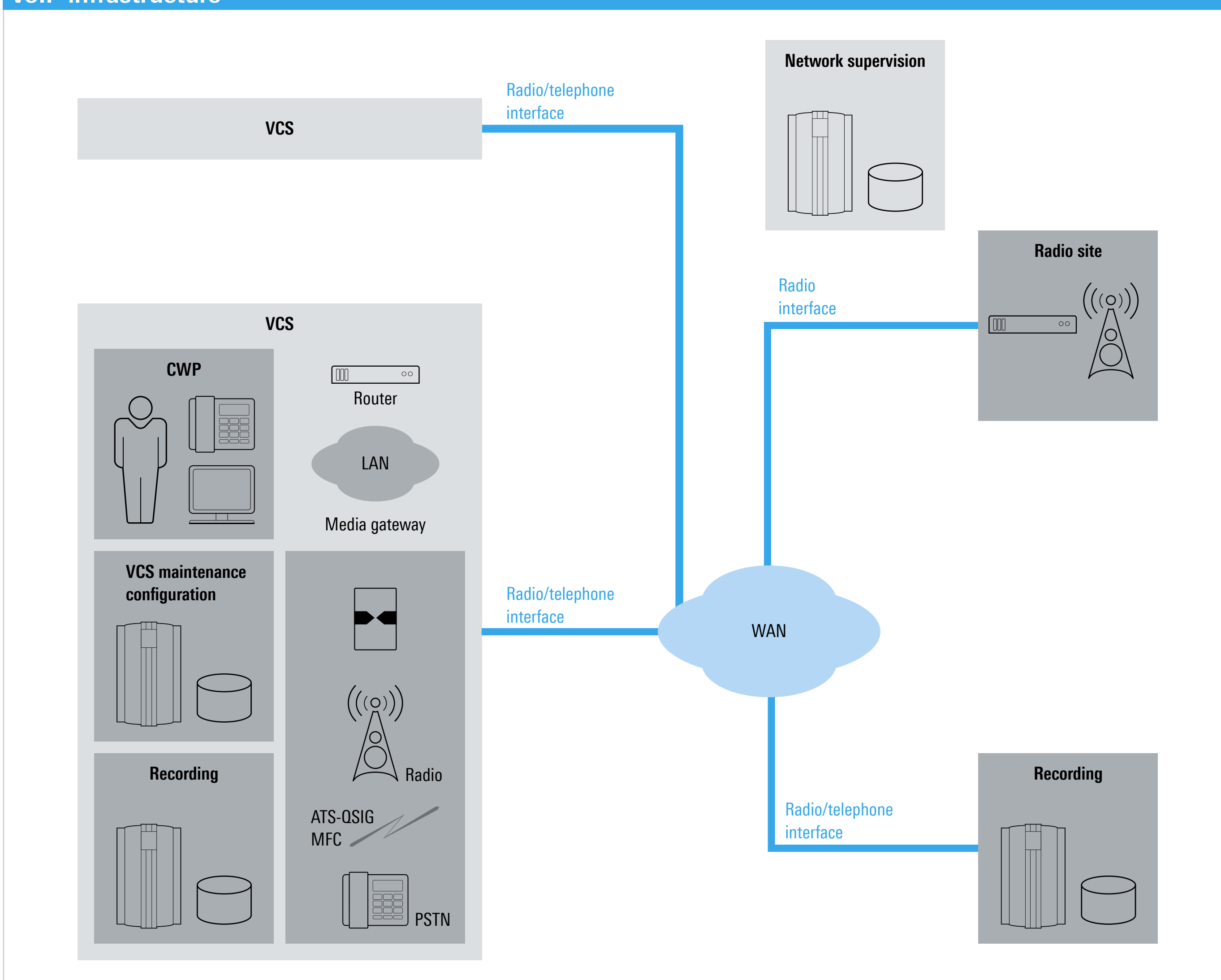

FIG 1 VoIP infrastructure for use in air traffic control (source: EUROCAE ED136).

interoperability. Today, every country has its own operational processes and different communications technology with which the air traffic controllers communicate with each other and with the pilots. Communications between air traffic controllers across borders run via telephone connections. Today, it is not easily possible for an air traffic controller in one country to optionally access the radio infrastructure of another country. But this is necessary if FABs are to be introduced, since an air traffic controller from one country must be able to operate the sector that is covered by the radio infrastructure of another country. Only VoIP technology provides the prerequisites for such functions. Plus, this technology offers the means of introducing additional performance features that make communications between air traffic controllers and pilots easier and more secure.

Standardization

In order to make VoIP usable for ATC applications, the European Organization for Civil Aviation Equipment (EUROCAE) created a working group – the WG67. Based on the existing requests for comments (RFCs), its task is to develop standards that will enable the use of VoIP. These standards allow interconnection of radios, VCS and CWP from different manufacturers. The WG67 consists of representatives from ATC organizations (air navigation service providers, ANSP) and industry. The infrastructure to be standardized is specified in the Vienna Agreement (FIG 1). The necessary requirements and specifications are defined in several documents [1 to 7] which were ratified at the end of 2008 and have been official EUROCAE documents since February 2009.

In April 2008 and March 2009, the specifications were validated in plug tests. For this purpose, the representatives from industry interconnected radios and VCS and performed various tests. Rohde & Schwarz actively participated in the plug tests by providing software defined radios of the R&S®Series4200 (see box on page 43), in which the VoIP functionality can be installed via software upgrade.

The tests showed that the EUROCAE standards are generally suitable for implementing VoIP-based VCS. The test results are also used to improve the interface specifications. In the next few months, new editions of the documents ED137 Part 1 and ED137 Part 2 are expected to be released.

The European Organization for the Safety of Air Navigation (EUROCONTROL) supports the migration of today's communications systems to VoIP systems with recommendations and guidelines to enable a uniform infrastructure across all participating EUROCONTROL countries. Furthermore, the EUROCAE standards are being introduced in the appropriate committees of the International Civil Aviation Organization (ICAO) so that they will be adopted worldwide.

The technology

The basis for the EUROCAE specifications are the RFCs developed by the Internet Engineering Task Force (IETF). The objective of WG67 was to remain as close as possible to the existing standards and expand them only whenever it was necessary to meet the requirements of the ED136 document. The basis for the VoIP-based interface between the VCS/CWP

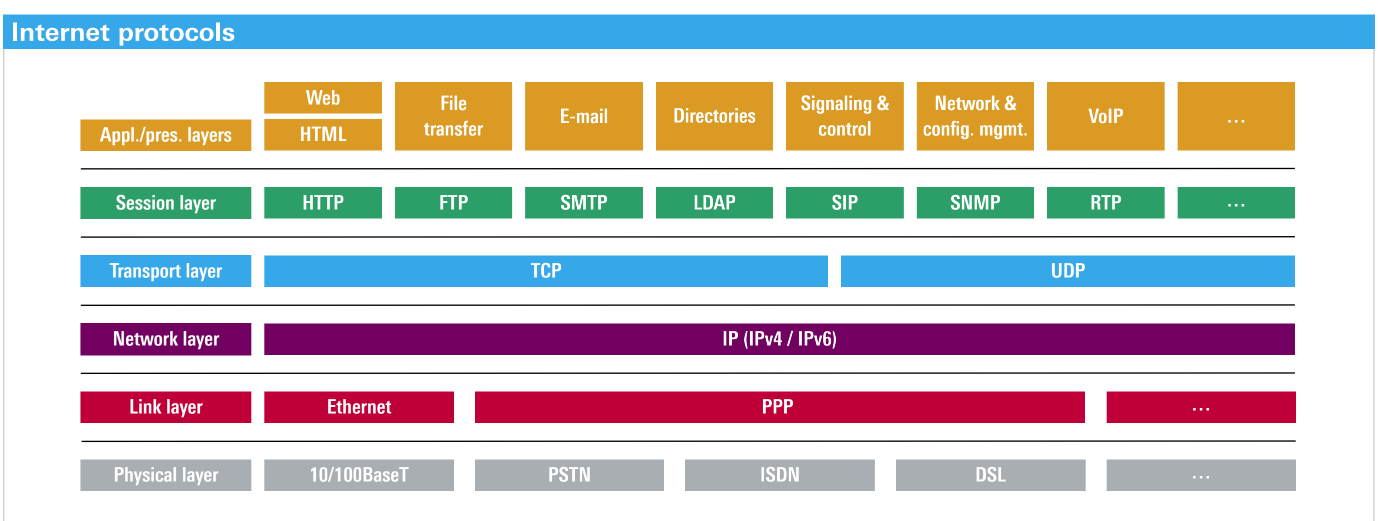

FIG 2 Selection of protocols commonly used in the IP world, presented in line with the ISO/OSI 7-layer model.

and the radios is the session initiation protocol (SIP) RFC 3261 and the realtime transport protocol (RTP) RFC 3550. FIG 2 shows the protocols that are common in the IP world, illustrated using the ISO/OSI 7-layer model.

The call setup is signaled between the radios and the VCS/CWP using SIP. For this purpose, the VCS/CWP sends the connection request to the corresponding radio, together with the parameters required for communications, e.g. the voice codecs to be used. The radio confirms the connection request, possibly with modified parameters, or rejects it with a cause value. Several VCS/CWP can set up an active connection to a radio, where the call setup is always initiated by the VCS/CWP.

After a successful call setup, a bidirectional RTP session is initialized between the radios and the VCS/CWP, where a separate session is set up to every VCS that sent a connection request. These connections transmit the voice packets. The packets are sent even if no voice is to be transmitted (no RF transmission), in this case without content. Besides voice, additional information for signaling is also transmitted in the RTP header. Since additional information must be transmitted together with voice in a VoIP system for ATM applications, the standard RTP header was extended. This extended RTP header transmits the push-to-talk (PTT) and squelch (SQ) signals. The transmitter is keyed with the PTT signal, and the SQ signal indicates the opening of the squelch at the receiver. In addition, signals for displaying the reception quality of the receiver are transmitted that enable the VCS/CWP to switch the best signal among several receivers through to the air traffic controller.

An important function that was implemented in the extended RTP header is the monitoring (keep-alive) of the connection between radio and VCS. For this purpose, the correct reception of the voice information is monitored by both sides. If no voice is transmitted, keep-alive signals are sent in the extended RTP header and their correct reception is monitored. Adjustable keep-alive timers are used to control the tolerance to interruptions and packet loss.

In most cases, the transmitters and receivers are positioned at different locations so that they do not interfere with each other. For this reason, VCS and CWP set up independent SIP/RTP sessions to the respective transmitters and receivers. The suitable audio streams are then combined in the VCS in each case to feed transmit and receive signals of the same radio channel to the headset of the air traffic controller. FIG 3 shows the phases of the VoIP call setup between a VCS and the transmitter or receiver of a radio channel.

VoIP implementation in the R&S®Series 4200

The R&S®Series4200 is the current generation of VHF and UHF radios for air traffic control and has been in use for about four years. The latest generation of this family of radios offers an additional digital interface in line with ITU-T G.703 as well as significantly higher processor performance and more storage capacity. As a result, the radios can be used not only for analog connections, as is still predominant today; alternatively, they can also be digitally connected to the voice switching system. For the connection via VoIP, the LAN interface that already exists for remote monitoring is used. All R&S®Series4200 radios that are already equipped with the

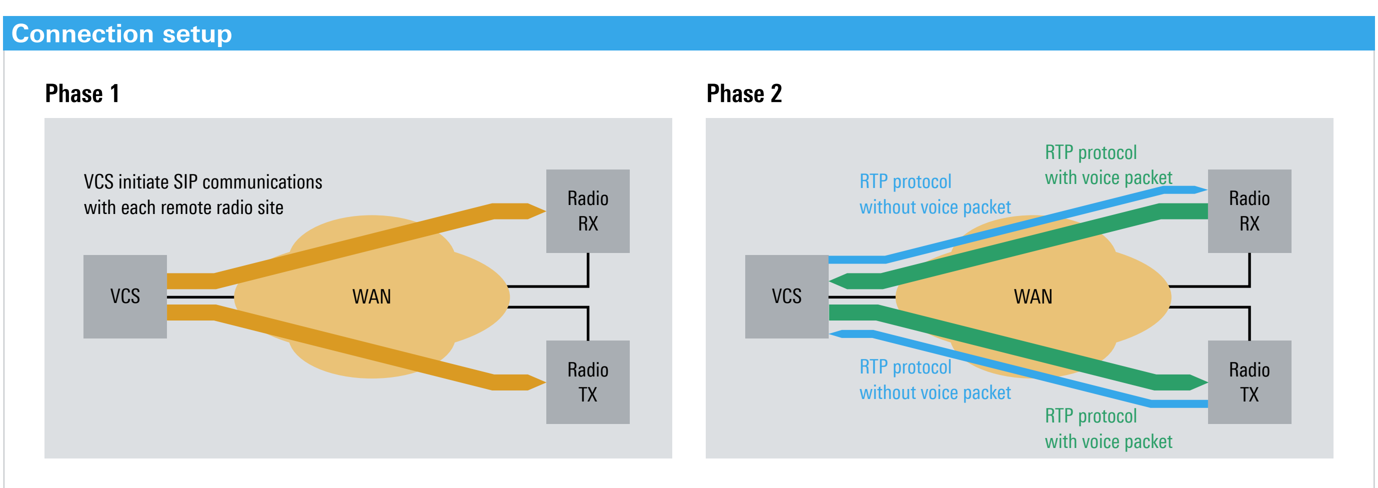

FIG 3 Call setup between VCS and transmitter/receiver of a radio channel via SIP and RTP (source: EUROCAE).

R&S®GB4000V and R&S®GB4000T with R&S®Series4200 via VoIP

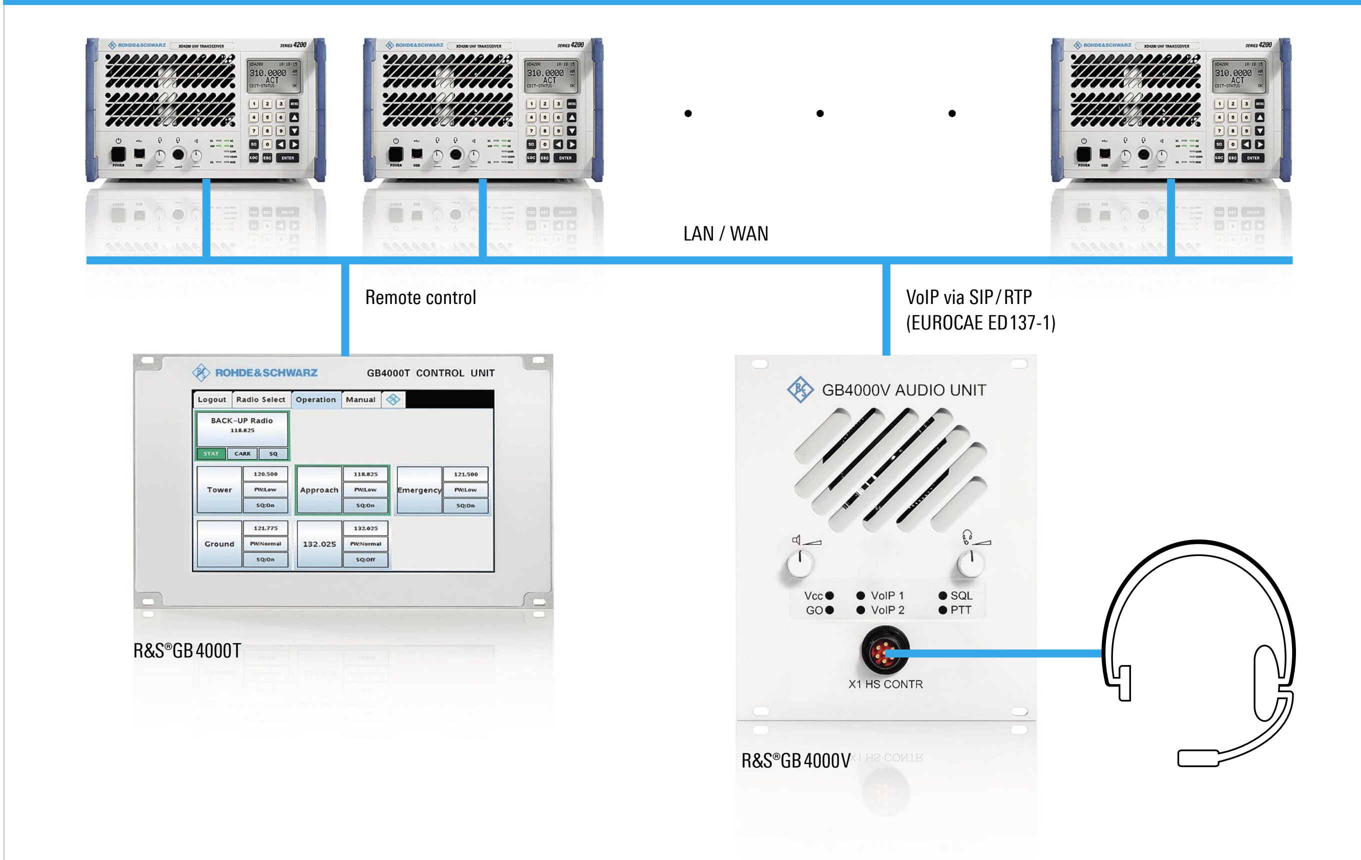

FIG 4 Implementation of a communications system via VoIP.

powerful processor can be upgraded to VoIP operation via software. As a result, they offer high safety of investment and certainty of planning, since they can be subsequently integrated into VoIP systems yet to be deployed.

For small systems with one or only a few controller working positions, Rohde&Schwarz offers additional components for VoIP-capable voice communications: the R&S®GB4000V audio unit and the R&S®GB4000T control unit. These system components make it very easy to implement VoIP-based communications systems, such as those required e.g. in a tower or for the apron check. FIG 4 shows such an application.

Summary

Voice over IP has proven its value in the classic telecommunications sector for several years now. With a few changes, which have been specified by EUROCAE in collaboration with representatives from ATC organizations and industry, VoIP is now ready to be used in ATC applications.

The R&S®Series4200 family provides the right radio for setting up voice communications systems that are open-ended for future needs.

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References
[1] ED136 "VoIP ATM System Operational and Technical Requirements"
[2] ED137 "Interoperability Standards for VoIP ATM Components – Part 1: Radio Interface"
[3] ED137 "Interoperability Standards for VoIP ATM Components – Part 2: Telephone Interface"
[4] ED137 "Interoperability Standards for VoIP ATM Components – Part 3: Recording Interface"
[5] ED137 "Interoperability Standards for VoIP ATM Components – Part 4: Supervision"
[6] ED138 "Network Requirements and Performances for VoIP ATM Systems – Part 1: Specification"
[7] ED138 "Network Requirements and Performances for VoIP ATM Systems – Part 2: Design Guideline"