VoIP in Air Traffic Management

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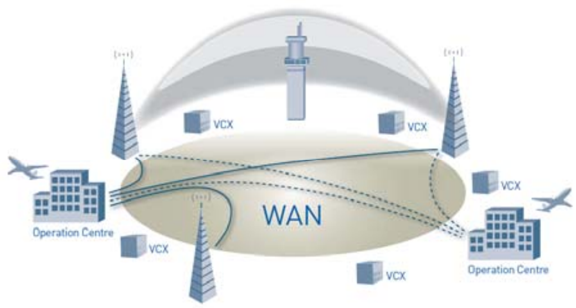
FREQUENTIS

ATC Networks are the application area for VoIP in ATM

- VoIP in ATM: Where will it be applied?
- A/G & G/G Communication:
  - Links between different ACC, Approach & Tower sites
  - Links between Controller and Radio equipment.



Agenda

- VoIP introduction
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Introduction

- VoIP means Voice over Internet Protocol . Aim of VoIP is to transmit audio data (especially voice) via IP packets.
- The VoIP network tries to combine the technology of the voice networks (PSTN, Public Switched Telephone Network) with the data networks (IP, Internet Protocol).
- This combination can be compared with the marriage of two very different personalities, with the hope, that a kind of synergy will result from this union.
- As you may already know by experience, to keep this alliance successful, it is essential to know the qualities and weak points of each person …
- The same rule applies for the marriage between voice and data.

Concept of connection oriented networks

- Usually voice networks (e.g. PSTN) are classified in the category of **connection oriented networks**.
- A channel is established between the source and the destination
  - By picking up the receiver the network gets the information that the user wants to use the telephone service.
  - The network sends a (free) tone and the user dials the telephone number.
  - When the destination pickups the receiver a point to point channel is allocated via the different switches used to establish the required path.
  - When both correspondents have hung up the communication is terminated, the used network resources are released and can be allocated for another communication

Concept of connectionless networks

- In contrast to the voice networks, data networks (e.g. IP) are usually classified in the category of **connectionless networks**
- In a data packet is included the source and destination address. This packet is sent into the network to be delivered to the destination address.
- The connectionless network can be compared to a postal transfer (the letter representing the data packet):
  - The letter is put in a mailbox, and if everything works well, the letter will reach it's destination
  - It is not predictable which exact path the letter will take to reach the destination.
  - Depending of the used path, the delivery time can vary considerably.
  - It is as well possible, that the letter may be lost or may be sent to a wrong address. As result the letter may never reach the final destination, or with a significant delay.
- For this reason connectionless networks are described as “best effort” or “non reliable” networks.

Why use VoIP ?

- Use IP for transfer of voice and data on a common unique network. This enables to reduce the costs, to increase the (audio) quality, and standard Internet equipment can be used:
  - Routers
  - Switches
- A quick adaptation is possible in case of failure of a router
- Resources are not blocked unnecessary

Protocols used in VoIP

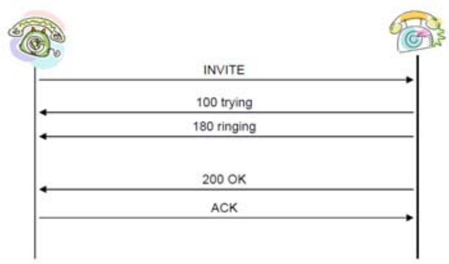
Media (Voice):

- Real time Protocol (RTP) -> UDP based

Signalling:

- Session Initiation Protocol (SIP) -> TCP based

Example of the SIP signalling of a successful VoIP call setup



Weak points of the IP protocol

→ Designed for a *best-effort delivery concept*

→ IP is considered as a non reliable protocol. Following topics are not treated by IP:

- the data corruption: there is no error detection available (function which is handled by the transport layer)
- the transmission delay
- the path used by the datagram.
- the incoming order of the datagram (the packets are not sequenced): a datagram A sent before a datagram B may arrive later at the destination than B.
- the loss or destruction of datagram.
- the reemission of datagram in case of non reception.

Requirements for VoIP

→ The quality of the transferred data is affected by following parameters: by the **flow rate** (downloading or broadcasting video signals), by the **delay** (for interactive actions or the telephony), the **availability** (access to a shared service) or by the packets **loss rate** (critical by downloads)

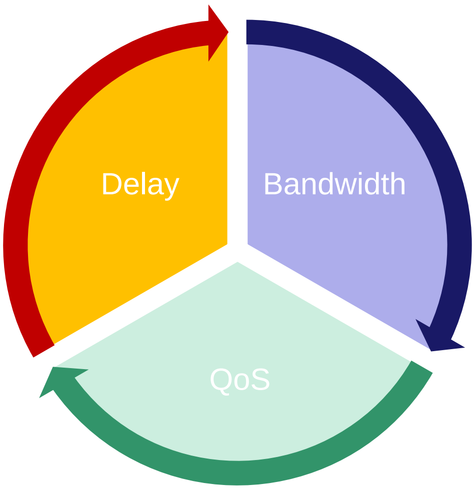
→ For VoIP the most important thing is the "**delivery on time**" of the packets.

→ Reliable transport methods aren't mandatory for the VoIP, and may even deteriorate the quality of the signal. For the VoIP it may be better just to handle losses, errors and/or congestions rather than trying to prevent them.

→ In case of loss of a TCP segment, TCP (which guarantees that the segments are received in the right order) will first resend the lost segment before sending the next one. This leads to a delay, which will increase by further losses.

Some thoughts around VoIP Network design

→ Performance Parameters for Network Quality



Performance parameters for VoIP

→ Following parameters have a strong influence on the psycho-acoustical quality of a VoIP communication:

→ **Loss of packets**

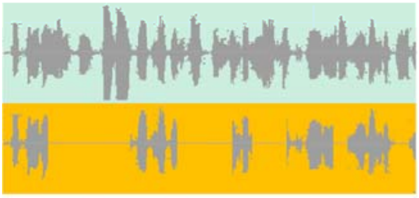
→ **Delay** (latency)

→ **Jitter** (temporary delay variation)

→ **Echo**

Loss of packets

- Packets may be lost due to
  - Corrupted packets (due to a transmission error) which are removed by the network equipment (no error correction on the IP layer).
  - Removed packets by the network equipment due to congestions (buffer full)
- The impacts of packets lost is minimised if:
  - The loss is randomly distributed over the time.
  - Short packets are sent (short voice samples ~10ms).
  - The lost packets are replaced or rebuild (no complete silence)



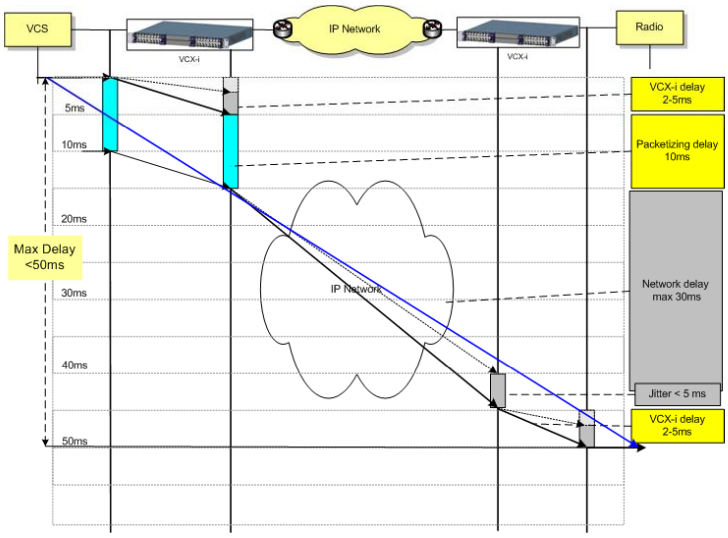
The maximum acceptable loss rate depends on the requirements given to the comprehension of the voice

Delays I

- Delays (static and variable) are generated over all the transportation path :
  - At the sender the static delays are generated by:
    - Packetizing (size of the audio samples influence delay).
    - Eventually compression (usually time intensive algorithms).
  - In the network the variable delays are generated by :
    - Waiting buffers in the routers.
    - The path used by the packets (number of intermediate routers)
    - Transmission time (usually negligible)
  - At the receiver the static delays are generated by :
    - The size of the jitter buffer
    - Eventually decompression.

ITU considers that a delay of :
- 0-150ms corresponds to a good quality communication
- 150-400ms corresponds to an acceptable quality communication

Delays II



Delay introduced by the network – What to do?

→ The network quality and the correct routing of the packets to their destination are the key requirements:

- The bandwidth must be wide enough, that even in case of high traffic the congestion risks are minimized
- The delay should be minimized as low and possible, but **most of all** keep them as constant as possible

→ The network must therefore offer a “**quality of service” (QoS)** able to handle better the available bandwidth and the waiting times.

Quality of Service (QoS) - Principles

- Two main category decisions:
  - Classification of the traffic in different classes (CoS):
    - Kind of data to be transferred ?
    - What are the different data classes?
  - Management of the different traffic classes :
    - How shall the classes be treated?
    - E.g.: different priorities, packets may be removed, …
- The classification is usually done by the routers at the end points of the network, but may as well be done by the terminals (e.g. FREQUENTIS VCX-IP).
- The management of the different classes is usually done by the central network routers.

Agenda

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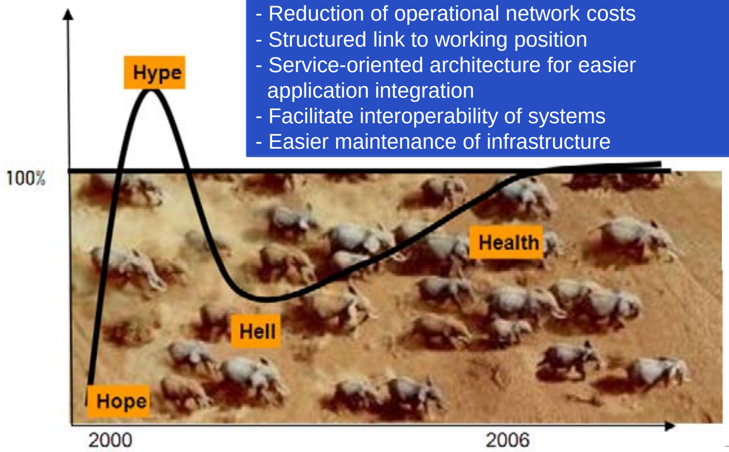
- Specifics of VoIP and Air Traffic Management
-

VoIP – from Hype to Health

Use IP technology where it brings benefits for the customer/ user (FREQUENTIS IP/TDM Hybrid approach):

- Reduction of operational network costs
- Structured link to working position
- Service-oriented architecture for easier application integration
- Facilitate interoperability of systems
- Easier maintenance of infrastructure



Advantages of IP Networking in ATM

→ Lower OPEX for telecommunication lines:

- TDM circuits are charged based on circuits (calls), and charging occurs whether the circuit is used or not
- IP communication is charged based on used bandwidth and actual data volume transferred, allowing for a dramatic reduction in telecommunication costs

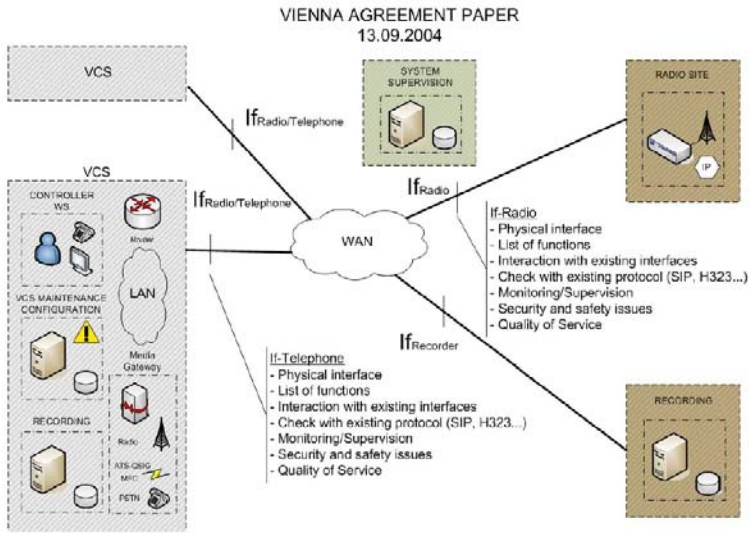
→ Future Proof:

- The communication industry is moving towards IP communication backbones, this will have impact on the availability and pricing of TDM and analogue links

→ The standards are now finalised:

- EuroCAE Working Group 67: VoIP in ATM, Released February 2009
- The ICAO WGI (Internet) is referencing to the EUROCAE WG 67 documents as part of the publication of ICAO standards for VoIP in aeronautical communication.

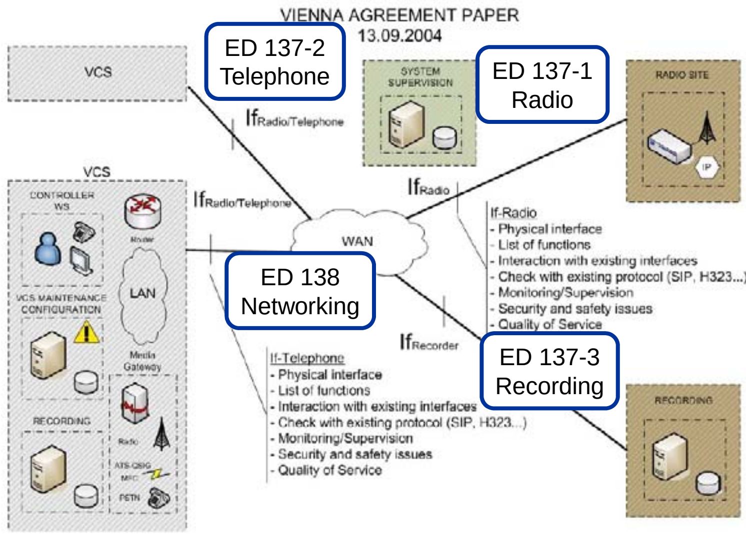
Eurocae WG 67 – VoIP in ATM



EC WG 67 - Deliverables

- **ED-136: Voice over IP ATM System operational and technical requirements**
  - to define and develop VoIP ATM System operational and technical requirements
- **ED-137 – Interoperability Standards for VoIP ATM components**
  - to define first, IP standards to fulfill operational and technical requirements, secondly to develop IP interfaces between IP components as defined in "Vienna Agreement"
- **ED-138 – Network requirements and performances for VoIP ATM Systems**
  - to define and develop network requirements and performances to fulfill QoS required to interconnect IP components
- **ED-139 – Qualification tests for VoIP ATM Components and Systems**
  - to define and develop validation requirements for VoIP ATM System and components

Eurocae WG 67 – Specification domains



Safety critical communication & IP Networks

Pros:

- Network availability: IP networking provides inherent routing capabilities, and is therefore less vulnerable to network link outages
- Business Resumption / Contingency: Planning and execution of resumption scenarios become more flexible and easier to accomplish by automatic re-routing and reconfiguring of network configurations

Cons and inhibitors

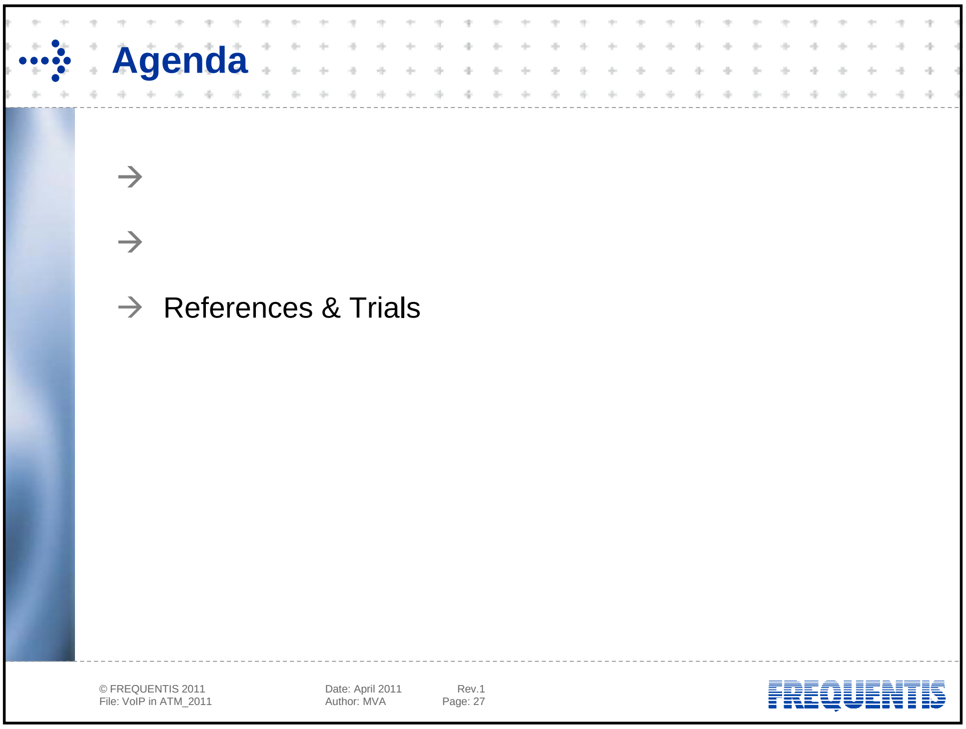
- Capacity and bandwidth planning: IP Voice communication is no longer planned according to number of voice channels, but according to Bandwidth and latency requirements
- QoS and priority handling: Mixing safety critical and administrative communication must be planned carefully as the network itself does not guarantee latency and packet-loss (especially in congestion scenarios)

Outlook 2011 for EUROCAE WG67

- Activities concerning VoIP in ATM are continued in the Voice over IP expert team (VOTE Group) – Frequentis leads VOTE Subgroup 2
- ICAO doc 9896 edition 2 refers to ED137A (radio and telephony interworking specification)
- Enhancement of EUROCAE standard currently under review concerning Radio CLIMAX operation over an IP network
- Further plug tests will follow in 06/2011 with a focus on Eurocae ED-137A – The updated version released in 2010
- FAA is holding a plugtest event in May 2011 with focus on the upcoming nextgen related upgrade of the voice communication infrastructure

Conclusion - Mature technology for safety critical use

- VoIP mainstream technology is well accepted and used in wide area telecommunication networks
- Standardisation for use of VoIP in ATM is finalised
  - ED-137 specifications are included in the upcoming 2nd edition of ICAO's Manual for the ATN using IPS Standards and Protocols (ICAO doc 9896).
- First deployments are being rolled out now
  - FREQUENTIS is the market leader for VoIP over IP in ATM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References & Trials
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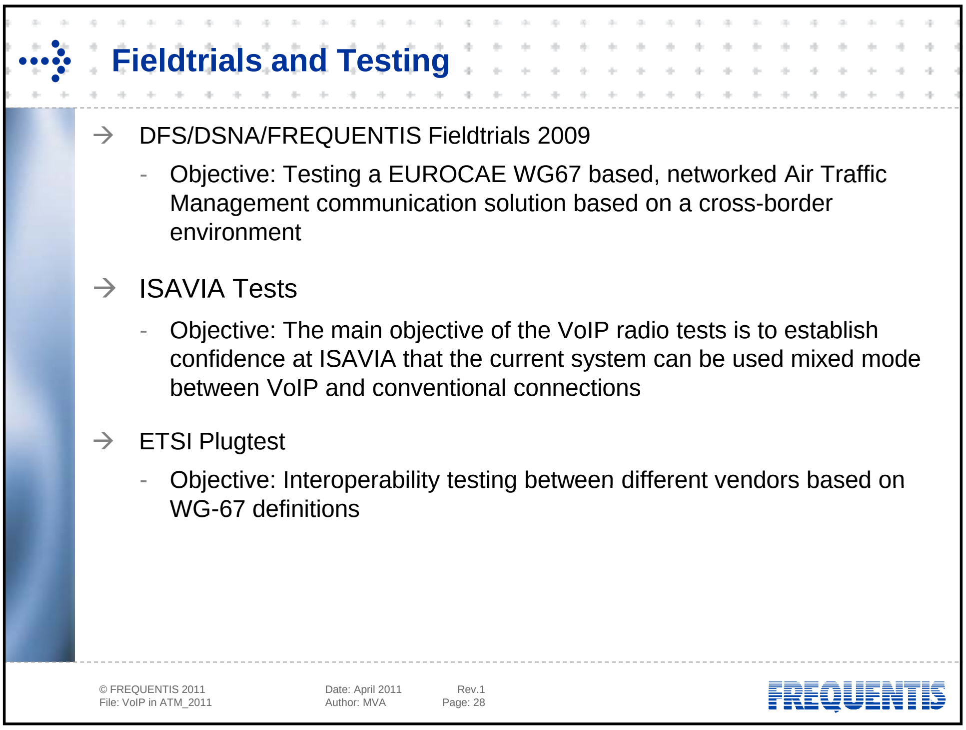
Fieldtrials and Testing
DFS/DSNA/FREQUENTIS Fieldtrials 2009
- Objective: Testing a EUROCAE WG67 based, networked Air Traffic Management communication solution based on a cross-border environment
ISAVIA Tests
- Objective: The main objective of the VoIP radio tests is to establish confidence at ISAVIA that the current system can be used mixed mode between VoIP and conventional connections
ETSI Plugtest
- Objective: Interoperability testing between different vendors based on WG-67 definitions
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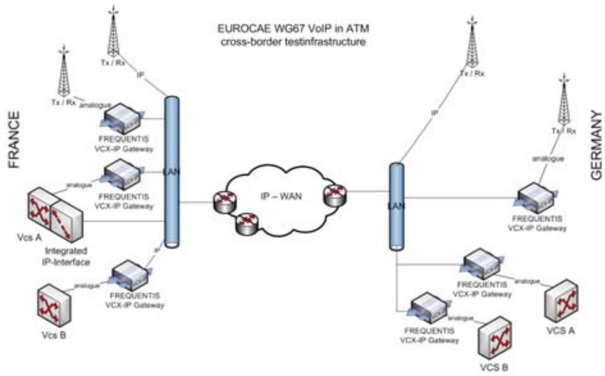
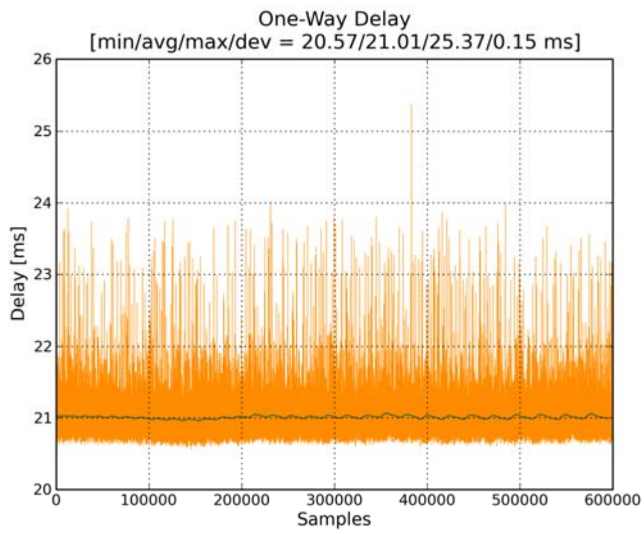
DFS/DSNA Field Trials - Topology

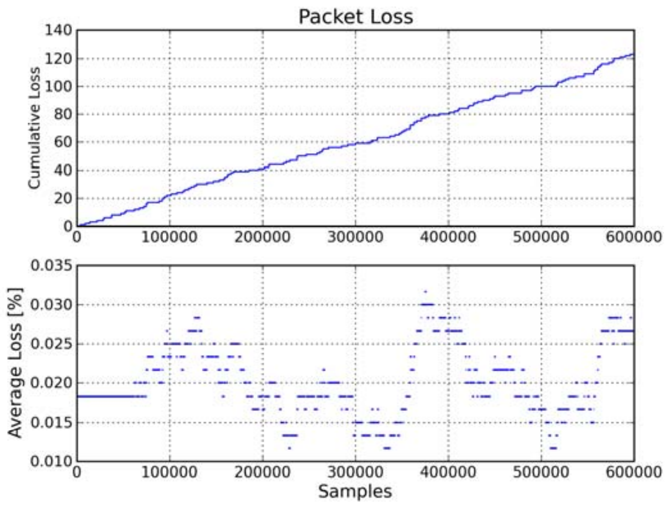
→ FREQUENTIS gateways connecting existing equipment to the IP network



DSNA/DFS Fieldtrials – Results



DSNA/DFS Fieldtrials – Results



VoIP trial in Iceland – Live Air/Ground Comms

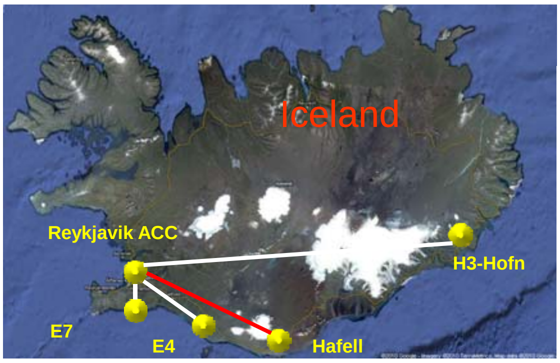
→ The main objective of the VoIP radio tests is to establish confidence at ISAVIA that the current system (VCS and associated infrastructure) can be used mixed mode between VoIP and conventional connections.

→ Participants:
- ISAVIA
- Northrop Grumman Park Air Systems
- Frequentis

→ Support from the Air:
- Air Iceland
- Icelandair

Geographical locations

- 5 Locations
- Reykjavik ACC
  1. IP Radio
  2. Analogue/IP GW connection
- Radio site Hafell
  - IP Radio
- Radio sites E4, E7, H3
  - Analogue/digital MUX connection



Best Signal Selection

- Goal: Measure the influence of VoIP Radio communication on the best signal selection voting mechanism
- Test Setup:
  - Multiple Receivers on different locations with both IP as well as TDM connections
  - Repeated communication between controller and pilot as the flight proceeds through the coverage area
  - Measure BSS voting outcome, perceived voice quality, mapped to location of flight.
- Outcome: Voting behaved as expected, i.e. no influence of IP radio communication on voting pattern.

ISAVIA Trial Conclusion

→ Best signal selection between radios using a hybrid IP/TDM infrastructure is behaving according to expectations

→ Voice quality is excellent, as long as the required QoS parameters are in place and respected

→ Interworking between VCS and Radio equipment according to EUROCAE WG67 Standards is performing and ready for use in an operational environment

1st Plug-Tests at ETSI

- April 14th – 18th 2008
- 5 VCS supplier
  - FREQUENTIS
  - CS (France)
  - Indra Sistemas (Spain)
  - Page (Spain)
  - S.I.T.T.I. (Italy)
- 1 radio supplier
  - SELEX Communications (Italy)
- Air traffic control agencies
  - Germany (DFS), France (DTI), USA (FAA), Europe (EUROCONTROL)

Plugtest #1 Results

→ Phase 1 VCS-VCS

Overall Results					
Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
103 (94.5%)	6 (5.5%)	11 (9.2%)	0 (0.0%)	109 (90.8%)	120

→ Phase 1 VCS-GRS

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
78 (97.5%)	2 (2.5%)	1 (1.2%)	0 (0.0%)	80 (98.8%)	81
Total:	80				

2nd Plug-Tests at ETSI + Field Trials

→ March 2009 (stage 1 and 2)

→ 7 VCS supplier

- CS , FREQUENTIS
  Indra, Page
  Park Air Systems
  Sitti, Topex

→ 4 radio supplier

- Park Air Systems, Rohde &Schwarz,
  SELEX Communications , Telerad

→ Air traffic control agencies

- ACG, DFS
  DSNA, EC
  FAA

Plugtest #2 Results

- Phase 2 VCS-VCS

Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
423 (95.7%)	19 (4.3%)	0 (0.0%)	38 (7.9%)	442 (92.1%)	480

Total: 442

- Phase 2 VCS-GRS

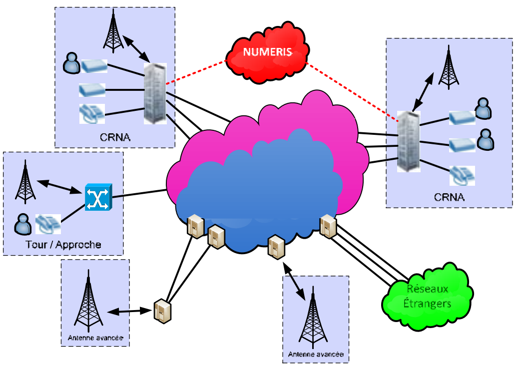
Overall Results

Interoperability		Not Executed		Totals	
OK	NO	NA	OT	Run	Results
483 (99.2%)	4 (0.8%)	70 (12.5%)	3 (0.5%)	487 (87.0%)	560

Total: 487

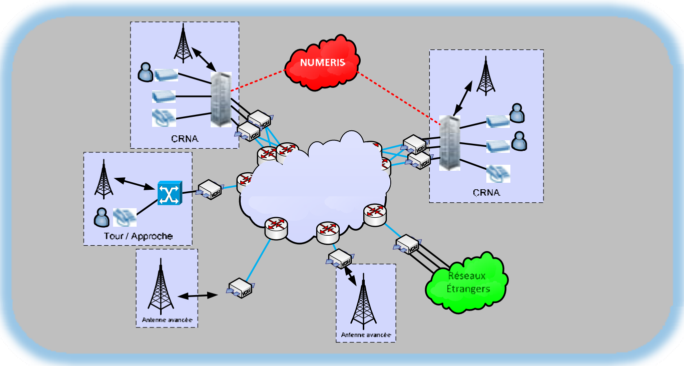
Deployment of VoIP Network in ATC: RENAR France

- Replacement of existing analogue network backbone by IP-backbone
- Reason: End of life announcement of analogue lines by 2012



Architecture of the IP based RENAR network

- Frequentis supplies the VCX IP Gateway components (iRIF and VCX IP Gateway) to connect the analogue radio and telephony services to the IP backbone
- The change of the backbone technology shall have no impact on the controller in the ACC or tower, and changes to the existing communications infrastructure (VCS, Radio) are minimal



Questions ?