Multi Network Voice Dispatching System

The unified Command Control and Communication solution Atos Trusted partner for your Digital Journey

Public safety, utilities, transportation and security organisations all are facing the challenge of managing their operational fleets at the highest level of security, reliability and effectiveness, regardless of network technology used.

However, meeting this challenge is possible only when a state-of-the-art communication control center enables these highly complex operations.

A control center has to provide a vast number of features to make the dispatching process as efficient and smooth as possible, give the operator intuitive access to all the necessary information and functionalities and it must be easy to operate.

Communication between the control center and field staff has to be reliable and always guarantee the required level of security.

Product Overview & Key Features

The Multi Network Voice Dispatching system is an application tailored to support operations of critical communication networks in the field of offshore windfarms, on and offshore oil & gas facilities, public sector and transportation.

Multi Network Connectivity

The Multi Network Voice Dispatching system relies on a full IP architecture which is capable of interconnecting to several different communication technologies such as TETRA, VHF Marine, VHF Air traffic, PSTN, PBX, GSM, LTE and many others. It provides not only the interconnection between such networks but also the capability of conferencing across those networks.

Scalability and Modular Design

Based on a Server / Thin Client architecture, the Multi Network Voice Dispatching system is capable of providing a large range of setups, beginning with a single work place configuration up to a complex multi user array of work places with distributed geographical locations. Several modules for interconnection to external networks and to other applications such as the Sea- and Site Surveillance system provide a flexible setup and configuration.

Reliability

The central server system can be provided in a fully redundant configuration which can even run on virtualized environments to meet the state of the art reliability and data center requirements.

Graphic User Interface

The graphical user interface is designed both for a touch screen and conventional operation.

Pre Integrated Interface

A variety of telecommunication equipment from different vendors are already preintegrated through their native IP interface as outlined below. In addition to this, the system is providing gateways for interconnection with almost any other technology via integrated Radio over IP Interface.

Special Offshore Capabilities

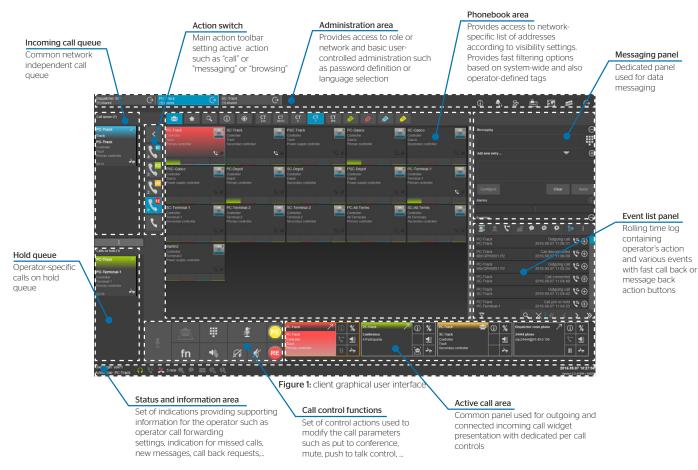
Being optimized for offshore operations, several related requirements are supported by the system such as interworking with dedicated DGMSS / DSC Marine VHF and Air traffic VHF, integration with Sea Surveillance and People Tracking System.

Interoperability with other subsystems

The Multi Network Voice Dispatching system interworks seamlessly with other subsystems such as Sea- and Site Surveillance system. This enables unique features such as "touch and call" e.g. to touch a target, such as a vessel, on the map and initiate a call procedure to that target.

User Interface

The Multi Network Voice Dispatching system client GUI enables quick and easy access to all communication resources independent of the communication technology and type of device. The graphical user interface is divided into several sections:



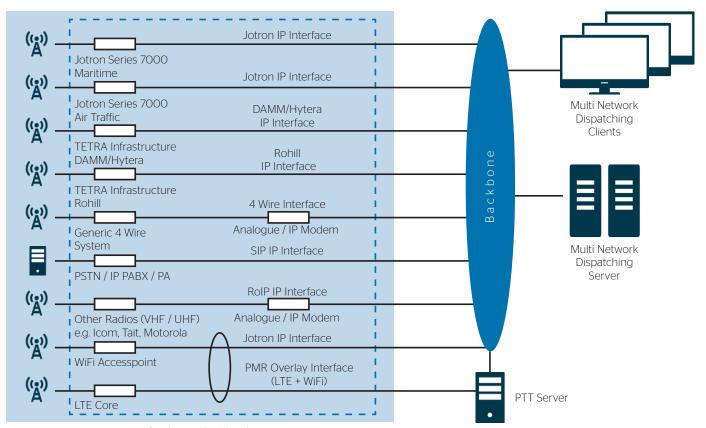


Figure 2: overview of available interfaces

Features and Functions

The following functionalities are available, provided the corresponding underlying communication technology is supporting them.

Generic Functions:

- Different login levels personal, parallel and shared roles
- · Role-specific phonebook
- Manual and automatic address tagging
- Audio routing adjustment
- · Local Voice Recording
- · Export of Voice Calls
- Export of Logs

Call Features

- Multi Network Connectivity (see Available Interfaces)
- In/Out individual full/semi duplex calls
- In/Out group semi duplex calls
- · Outgoing Broadcast Calls
- Ambience Listening Calls
- Conference Call
- Split Conference
- Active Conference Bookmarking
- Call Focus
- Speech Item Request
- Speech Item Pre-emption

Messaging

For TETRA:

- In/Out individually or group addressed SDS
- In/Out individually or group addressed status
- Emergency Status
- Call Back Request
- Man down and Clear Man Down
- SDS Message Template

For GSM:

- In/Out individually addressed SMS
- SMS Message Template

TETRA Supplementary Services

- · Talking Party Identification
- Group Mute/Unmute
- Call Transfer
- Call Forwarding
- DGNA (*)
- Temporarily Enable / Disable (*)
- Monitoring Service (*)
- Discrete Listening (*)

The Multi Network Voice Dispatching system is pre-integrated for the native IP interface of TETRA base stations from DAMM and Rohill. In case other TETRA vendors have to be integrated or no IP connectivity is available, a radio link for interconnection to the corresponding TETRA infrastructure can be used. However due to lack of standardizations in TETRA, this radio link does not provide the full feature set in such cases. Features not available with a radio link are marked with (*) in the above table.

Available Interfaces

- Generic: IP, PSTN, PBX: SIP RFC 3261
- TETRA: IP Interface to TETRA Flex DAMM and IP Interface to Rohill TETRA
- VHF-Marine and VHF Air Traffic: IP interface to Jotron 70xx Series
- Radio over IP (RoIP) with TETRA PEI tunnel for remote operation of TETRA Modems
- RoIP Interface: for remote operation of various technologies
- SMSC: IP interface for bulk SMS Messaging
- LTE Overlay via dedicated PTT Server

Accessories and Audio Equipment

External loudspeakers, gooseneck microphone with dedicated pad from TiproFoot PTT Switch, headset

Required Hardware and Operating Systems

Client PC: Personal PC, CPU Intel Core Duo or equivalent, memory 4GB+, sound card, Windows 7/8/10

Server Machine: Off-the-shelf Intel-based motherboard, CPU and memory depending on scalability requirements, OS Ubuntu 14.4.x

Optional: RoIP gateway hardware module translating SIP-based communication to:

- Analog voice and PEI/AT commands based signaling
- Analog voice and serial port based squelch control

About Atos

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