A Multimedia over IP Integrated System for Military Communications

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ABSTRACT

The leading role of a common IP-based infrastructure, able to sustain the increasing communications and information superiority, requires the use of a flexible and dynamically-matching communications architecture, especially when the new concepts of Network Enabled Capabilities and Network Centric Warfare are taken in consideration.

Modern technologies make it possible to enhance the Network role, as an integrator of platforms and systems. With the NEC approach much attention is however paid to maintain the assets of the already existing legacy systems. It has to be stressed how communications requirements raise dramatically now, either in quantity either in quality, in order to effectively improve sensor capabilities, decision tools, weapons lethality and information warfare. This new approach is therefore essential to grant superior decision making, power projection and protection.

To operate in a such scenario, an Integrated Telecommunications System must be designed according to three imperative features:

- *native IP(v4 and v6) based system: Internet Protocol represents the common network environment to enable interoperability and information sharing among nations;*

- *compliance with legacy communications technology: old-technology entities are expected to survive for a long time also in a modern battlefield;*

- *compliance with sophisticated architecture for QoS provisioning: bandwidth may be extremely narrow and expensive, especially in critical connections (Satellite Access Networks, Capillaries in the battlefield, ...). The best-effort approach is definitively inappropriate for a real battlefield scenario.*

A prototype of an Integrated Telecommunication System is here depicted. It is the "SIT di Mariteleradar", a project led by the Istituto Vallauri of the Italian Navy. As the above mentioned features were inserted in the very beginning of the design process, SIT is now effectively striving to meet the required capabilities that are necessary to adhere to the innovative NEC concept.

1.0   INTRODUCTION

In the paper we will give an overview of SIT. In particular we will describe the architecture of a system, which directly derives from ITU-T H.323 standard framework. The choice of this standard was motivated by the great flexibility it deals in the development of new services and by its capability of integration with legacy communication technologies. SIT implements a Gatekeeper, according to ITU terminology, acting as AA server (Authentication and Authorization) and signalling centralizer. The Gatekeeper Routed model allows a fine control of the communication activity present in the network and is a requirement for a flow control. The gatekeeper, acknowledged the need of two or more entities to communicate each other, turn its request of bandwidth to the dynamic signalling inter-working unit which colloquists with a Bandwidth Broker in the Transport Network. Featuring with dynamic network architectures and bandwidth optimization, our integrated communication system permit direct communications between entities, consequently to the signalling process and the establishment of QoS policies when supported by the core network. This prevents gatekeeper to act as an application proxy and to increase robustness to the system. SIT supports multipoint communications implementing two distinct Multipoint Control Unit integrating itself the Multipoint Controller and the Multipoint Processor: Int_MCU and Ext_MCU. The last is dedicated to the interaction within legacy entities (radio devices not supporting the IP protocol). The communications with those devices is gathered to a Radio Gateway H.323v4 compatible. In the paper we will describe the way to add functionalities to endpoints, needed by the system compliance with legacy radio vectors (Push To Talk, ...), adopting the ITU-T H.450 Supplementary Services Framework. To prevent any possible interoperability conflict with other standard H.323 endpoints, the system develops services without modification to the under layering message structures.

Coming back to the main target of SIT, which is to innovate communication technologies in a way to engage Innovation required by the Network Enabled Capabilities model, some more consideration about the Network architecture itself needs to be made. To Achieve information superiority translatable into combat power it is fundamental to create a common infrastructure assuring communications at strategic and tactical level. The network needed must always provide a critical level of service, appropriate to the type of service provided. In this paper we focus on a main network with Differentiated Services according to IETF proposal, together with policy based network management and bandwidth brokerage with the capability to perform flow admission control based on network resources availability. We will stress the role of SIT and the characteristic of the Gatekeeper to act as the Application Server in the access network capable to exchange information within the Network Management System of the core network with functionalities of bandwidth brokerage.

We will then scale considering the different issues related to a Differentiated Services over Multi Protocol Label Switching core network. In this approach we suppose that a service architecture based on MPLS has been implemented in the transport network, but differentiated services must also be supported in order to assure that packets marked with various Differentiated Services Code Points receive the appropriate QoS treatment at each Label Switched Router.

Finally we will describe the dynamic signalling inter-working unit interacting with the gatekeeper. A customized version of communication protocol like COPS (according to the language used by the bandwidth broker) is adopted. Thus optimizing for managing the multimedia flows when requesting access to the DiffServ architecture so making able the Gatekeeper to commit resources for the calls that is managing. The procedures needed to establish the QoS call over a multi-domain DiffServ scenario will be

also discussed; the needed procedure are dynamically started at call set-up time avoiding the users from drafting unsatisfactory static Service Level Agreements.

2.0   SIT: AN IP-BASED INTEGRATED TELECOMMUNICATIONS SYSTEM

2.1   SIT Technology Concept

The NC3TA [1] establishes guidelines inside all the initiatives aimed to provide a multiple-network infrastructure to military forces, as in Network Enabled Capabilities - NEC, Network Based Defence - NBD, Network Based Operations - NBO, Network Centric Warfare - NCW, Network Centric Operations – NCO. The main concepts of NC3TA can be summarized as follows.

- **(C1) Robustly Networked Forces**: the goal of this network enabled environment is to grant protected, assured and interoperable communications;

- **(C2) Information Sharing**: since the Network environment will include assets of many different nations and with different levels of classification, the Network has the high complex issue of sharing all available information that is relevant to the mission;

- **(C3) Dynamics and Flexibility**: a robust and flexible architecture is necessary to ensure that the information and the related infrastructures are reliable, secure and adaptable enough to meet the dynamically changing mission needs;

- **(C4) Inclusive and Flexible Acquisition**: the successful achievement of the Network environment will require the coordination of the procurement processes across NATO, the member nations, and industry. This coordination will promote the rapid insertion of new technologies, facilitate coherence between acquisition programs, and provide an incremental approach to deliver and maintain NEC-ready components. The sheer number of computing devices and networks in a net-centric environment will probably lead to increased reliance on Commercial off the Shelf (COTS) based equipment and services to manage costs, availability of technology, and length of development cycles.

The NEC scenario is composed by a set of networks that can significantly differ both from technical (e.g. strategic high speed networks, tactical ad-hoc radio, wireless networks, etc.) and logical (e.g. different levels of classification, diverse nations, etc.) perspective. Furthermore, these networks could dynamically change (e.g. in the battlefield). In this complex framework, all different networks must be internetworked in the most efficient and effective method. In this environment, the TCP-IP service model is fundamental, considering its unique ability to handle heterogeneity. Acknowledging the TCP-IP suite as the nerve centres of a Network Enabled Capability, our efforts focused on the development of a IP-based Integrated Telecommunication System for Voice and Video Communications between users who are joining different networks.

The design of SIT [2] relies on these three pillars:

- **native IP(v4 and v6) based system**: Internet Protocol represents the common network environment to enable interoperability and information sharing among nations;

- **compliance with legacy communications technology**: old-technology entities are expected to survive for a long time also in a modern battlefield;

- **compliance with sophisticated architecture for QoS provisioning**: bandwidth may be extremely narrow and expensive, especially in critical connections (Satellite Access Networks, Capillaries in the battlefield, ...). The best-effort approach is definitively inappropriate for a real battlefield scenario.
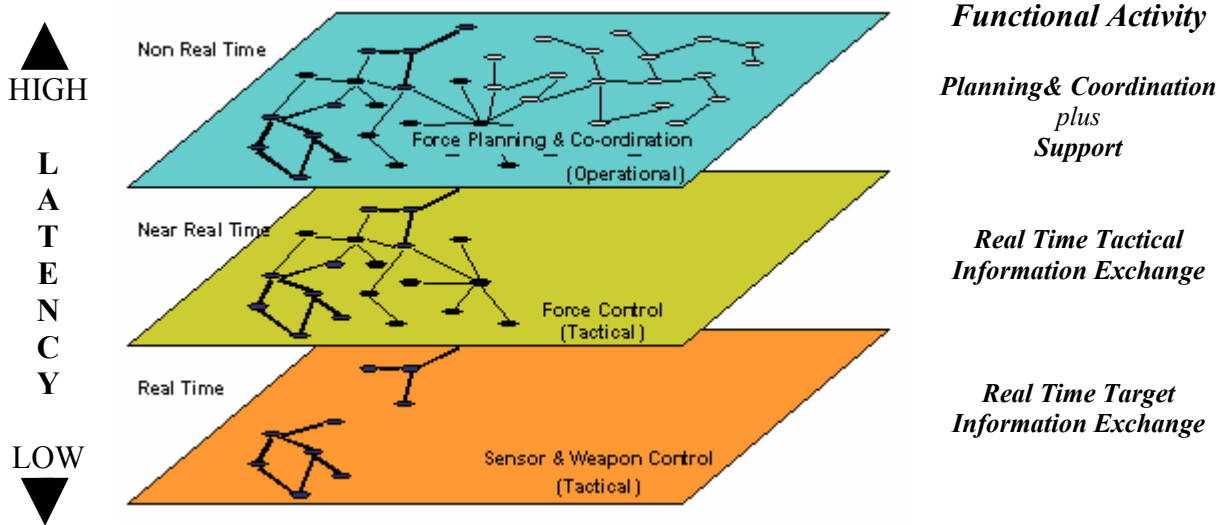
Figure 1: Global Network Scenario

SIT services are usable in the following Functional Activity, as it is shown in figure 1:

- **Real Time Tactical Information Exchange**: Link and Voice Radio;

- **Planning and Coordination**: Satellite and terrestrial radio, national command support networks;

- **Support**: National Networks and Internet.

Since SIT is developed on IP, it should be considered as a set of new multimedia services that are directly available in the network; no change has to be inserted in the under layering technologies. This approach prevents from ad hoc solutions for particular required features. As an example, security issue could be examined: a network level framework like IPSEC, already adopted for data exchange, is promptly deployable for such requirement. SIT adheres to all concepts of NATO NEC, as summarized in the NC3TA:

- (C1): SIT represents a services framework enabling communications between nations;

- (C2), (C3): With inherent Voice and Video Capability over the TCP- IP suite, SIT exploits the dynamics and flexibility of IP protocol. A simple human interface collects information from different scenarios and media, thus preventing the use of different devices for data and voice;

- (C4): SIT is compliant with International Standards and evolves from wide adopted Open Source Projects. Effective interoperability between different producers of standard equipments is therefore matched with full interoperability with COTS and civilian Networks.

2.2   Why an Open Environment is paramount for an effective design

To meet the challenge, SIT has been designed to grant the following features:

- ability to serve as "common environment" for different COTS devices, provided by different suppliers with different protocol solutions;

- strong commitment to interoperability, avoiding any vendor lock-in mechanism;

- ability to promote the rapid insertion of new technologies, without reducing operational readiness;

- high scalability to match different possible deployments, from ship-to-station connections to task group communications;

- flexibility to meet the network architecture where it has to be hosted;

- prolonged maintainability, without any dependence over COTS elements (i.e.: operating systems) that are susceptible to an interruption of vendor assistance;

- fully accessibility of any part of the implementation, either technically, or legally, to grant the customer the leading role to promote any improvement.

In order to meet these commitments, the easy way to adopt commercial solutions had to be dropped out; many vendors are already presenting their multimedia IP approach as COTS solutions, but interoperability and flexibility are here a major issue.

To grant good interoperability and to avoid lock-in vendor mechanism, it is necessary to adhere to open standards; many suppliers pretend to fully respect open standards, but slightly personalizations are often performed ("*adhere, enhance, disrupt*" policy). These extensions are advertised as "enhancements", but such behaviour leads to progressive incompatibility with competitors products.

On the contrary, a full open standard compliance assures a common basic set of functions from any product, no matter which vendors sell it. On such a common platform, further extensions are possible.

Such a policy can effectively promote a flexible environment where to insert COTS items, but it is not enough to grant the customer with the full control of the development process. To acquire such ability, it is necessary a further step: the adoption of open source implementation, where any software details can be fully disclosed and modified without patent harassments.

Thanks to adoption of free software to develop all the SIT modules, it is now possible to full control maintainability and improvement; there are no risk that a vendor policy could interfere with SIT adoptions. On the contrary, if a proprietary COTS platform were adopted, it would now be necessary to face the policy of the supplier, who probably has no interest to keep supported the same operating system for more than 4-5 years. Dropping assistance forces customers to buy a newer release. Such vendor policy is particularly effective in a COTS network environment, where fully compatibility with evolving devices requires a up-to-date drivers collections.

To choose free software is the only viable approach for the customer to maintain full control about what has to be updated. It is not a licences issues, it is a configuration issue; each time a software is upgraded, a downward compatibility has to be maintained. With free software adoption, such effort is much easier.

The maintainability issue is paramount for a system that has to be adopted in a governmental environment, where a 20 years life-cycle is a normal assumption, that could not be granted with proprietary COTS software designed to be replaced each 4-5 years.

For SIT, all the project has been led by the Istituto Vallauri of the Italian Navy; a large amount of software has been obtained from Internet, browsing many different well documented free projects as the OpenH323 project [3], the GNU Gatekeeper [4], the freeRADIUS Server project [5], etc ...

2.3   Brief Description of ITU-T H.323 Framework

The core system enabling the Integrated SIT platform is built adopting the H.323 standard. H.323, a protocol suite defined by ITU-T, is intended for voice transmission over internet (Voice over IP or VoIP). In addition to voice applications, H.323 provides mechanisms for video communication and data collaboration, in combination with the ITU-T T.120 series standards. H.323 is one of the major VoIP standards, just as Megaco and SIP.

H.323 is an umbrella specification, because it includes a various other ITU standards. The components under H.323 architecture are terminal, gateway (GW), gatekeeper (GK) and multipoint control unit (MCU).

Terminals represent the end device of any connection. They provide real time two way communications with another H.323 terminal, GW or MCU. This communication consists of any possible combination of voice, video and data.

Gateways establish the connection between the terminals of the H.323 network with those belonging to networks with different protocol stack such as the traditional PSTN network , SIP or Megaco.

Gatekeepers are responsible for translating between H.323 and IP addresses, where H.323 addresses can be E.167 telephony numbers or H.323 aliases. Gatekeepers manage the bandwidth and provide mechanisms for terminal registration and authentication.

MCUs take care of establishing multipoint conferences. They perform a mandatory Multipoint Control for call signalling and conferencing and an optional Multipoint Processor, intended for switching/mixing of media stream. MCU provide also real-time transcoding of received audio/video streams.

2.4   The Choice of VoIP standard for Communications

Though the choice among H.323, SIP and proprietary protocols seems a purely technical one, it has implications on the interoperability with future expansions, inter-department trunking and the deployment of new advanced features, like messaging, etc.

When referring to VoIP the main focus is on voice services, which may be misleading regarding the support of video. VoIP standards do have the capabilities of signalling and are able to initiate multimedia communications. This scenario details how VoIP technologies / standards and videoconferencing solutions may be seamlessly integrated. The goal is to provide the users with a global architecture derived from VoIP standards, giving videoconferencing systems the chance of becoming widely used and adopted. Videoconferencing systems have the purpose of facilitating meetings of remote participants, and to support the illusion that they are all sharing the same space and communicating as if they were in the same room. Perfect videoconferencing sessions are achieved when the technology is no longer noticeable. Even though the perceived quality of video and audio plays the most important role, there are a number of other factors influencing the perception of successful videoconferencing:

- accessibility of the system: the system should be accessible on a broad area giving users the easiest way of communicating without worrying about how to join a conference

- value-added services: data/application sharing and voice mail are just two examples of value-added services which are not feasible with classic telephony systems, yet they may improve the quality experienced by the user;

- interoperability among different technologies: the system should be transparent to different technologies in order to give the users the chance of having seamless connectivity.

In order to describe the possible integration of advanced services we need to examine which are the possible applications related to the Videoconferencing scenario. Basic use of videoconferencing systems is useful to support functionalities, planning and coordination, but is not common on tactical links (mostly because of the rare bandwidth available); more specific applications may therefore be developed on top of the basic functionality, with enhanced options.

Possible military applications could be GPS plot diffusion, common picture distribution, tactical order transmission, or similar activities.

Significant civil applications could be considered:

- Telecommuting - Telecommuting is a broad term referring to corporate employees who interact electronically with corporate resources and people, or to home-based consultants communicating with inter-company business partners.

- Telemedicine - Videoconferencing solutions delivering high-quality video images to remote medical specialists. Specialized videoconferencing devices may be required to enable high-quality video contents not available with the standard videoconferencing systems.

- Distance Learning - Video lectures, remote guest speakers invitation to a classroom and private lessons to groups of students located in different locations

- Customer Services - Videoconferencing-based customer services enable call center operators to be more effective while interacting with customers.

- Virtual/Remote Laboratories - Remote laboratories allow researchers to share advanced appliances using existing network infrastructure.

On the application side, a successful integration scenario would require service specific devices for content-delivering to the end-user: for basic meetings, simple desktop conferencing equipment may be enough. On the technical side, this scenario would require servers (to build a global architecture accessible by all group users), gateways (to provide interoperability with different access technologies and different IP telephony protocols), conference bridges and multipoint conferencing units (to provide capabilities for multipoint conferencing).

3.0 "SIT DI MARITELERADAR" OVERVIEW

A relevant characteristic of the integrated telecommunication system designed by Istituto Vallauri is its ability to interact with military radio not supporting IP protocol. In section 3.1 we will focus on the development of standard H.323 new services for implementing peculiar signalling functions needed by radio device. We will stress how the adoption of the H.450 service creation framework permits interoperability with COTS standard devices. In the rest of the section, we will give an architectural overview of SIT, describing its main functionalities.

3.1 SIT innovation

H.323, briefly described in section 2.3, is a standardized signalling protocol for Voice over IP (VoIP) networks, which defines the terminal equipment and services that enable real-time multimedia (data, voice, and video) communication over packet-based networks. One of the major advantages of carrying voice over Computer Networks, as opposed to the traditional Switched Circuit Networks, is the possibility of creating a wealth of services and integrating them into the network with minimal time and cost. A subset of these services need to interact at the signalling layer and H.323 (version 5) provides mechanisms for this aim.

New signalling services imply the transport within existing H.323 messages of information, necessary to their operation. Obviously, we can not simply integrate additional information into existing H.323 messages whenever we implement a new service, because this would result in applications that are non-standard, non-interoperable and quite simply useless. As a result, signalling service creation would be impossible in H.323, if some mechanisms were not standardised for this aim. Fortunately, this is exactly what H.323 provides: a set of protocol extension mechanisms that we may use to create signalling services by adding information to the underlying H.323 messages. This information is added in such a way that the high-level structure of the messages is not altered, but rather cleverly integrated into a series of framework structures already present in the messages. Using these mechanisms we can create new H.323 services and

at the same time maintain interoperability, which is one of the major reasons why H.323 is so popular in VoIP networks.

Using these extension and service creation mechanisms, we created our own signalling service. This service, which we have called MCU RePort (MCU-RP), and the way in which it uses H.323's extension mechanisms are the focus of the following section.

H.323 defines three mechanisms that make the protocol both extensible and flexible, offering the ability to create new services or customize existing ones. H.450 is the primary service creation framework, while H.460 (Generic Extensibility Framework) and H.323 Non-standard Parameters are extension mechanisms. All these mechanisms work by providing hooks for implementers to add their service and extensibility information into the messages in a transparent manner. H.323 is a binary protocol and, like most other binary protocols, it requires an abstract notation for defining its message structures so that programmers don't have to work at the binary layer. ASN.1 (Abstract Syntax Notation Number One) is the abstract notation used by H.323 and the ASN.1 modules used in the MCU-RP protocol is described in section 3.2.1.

The other innovation was the implementation of a middleware granting the interoperability between the "new" technology (VoIP), value added services and the former telecommunication radio facilities (classical HF,V-UHF radios). This was accomplished by means of a Radio Gateway capable to understand the H.323 enhanced protocol and to translate these messages in order to pilot an electronic radio enabler. This radio enabler is a kind of signalling transductor to drive radio equipments, enabling users from the IP side to access the radio-link communication towards the external radio correspondents.

3.2   SIT Architectural Overview

According to ITU terminology, the Gatekeeper acts as AA server (Authentication and Authorization) and as a signalling centralizer . The Gatekeeper Routed model allows a fine control of the communication activity in the network and it is a requirement for a flow control. The SIT supports multipoint communications implementing two distinct Multipoint Control Units: Int_MCU for Internal conference support and Ext_MCU for external radio channelling.

3.2.1   Use of H.450.1 generic functional protocol for the support of supplementary services in H.323

The ITU-T H.450 Supplementary Services Framework is adopted to add functionalities required by compliance with legacy radio vectors. The underlying H.323 message structure is preserved to avoid any conflict between legacy radio elements and network devices.

The SIT multipoint conference service is a relatively simple service that could be used to explore and demonstrate the functionality of H.323's extensibility and service creation frameworks. Through the implementation of this service, we have shown that the H.450 Supplementary Services Framework provides highly flexible mechanisms useful to create reliable signalling services within H.323 compliant environments.
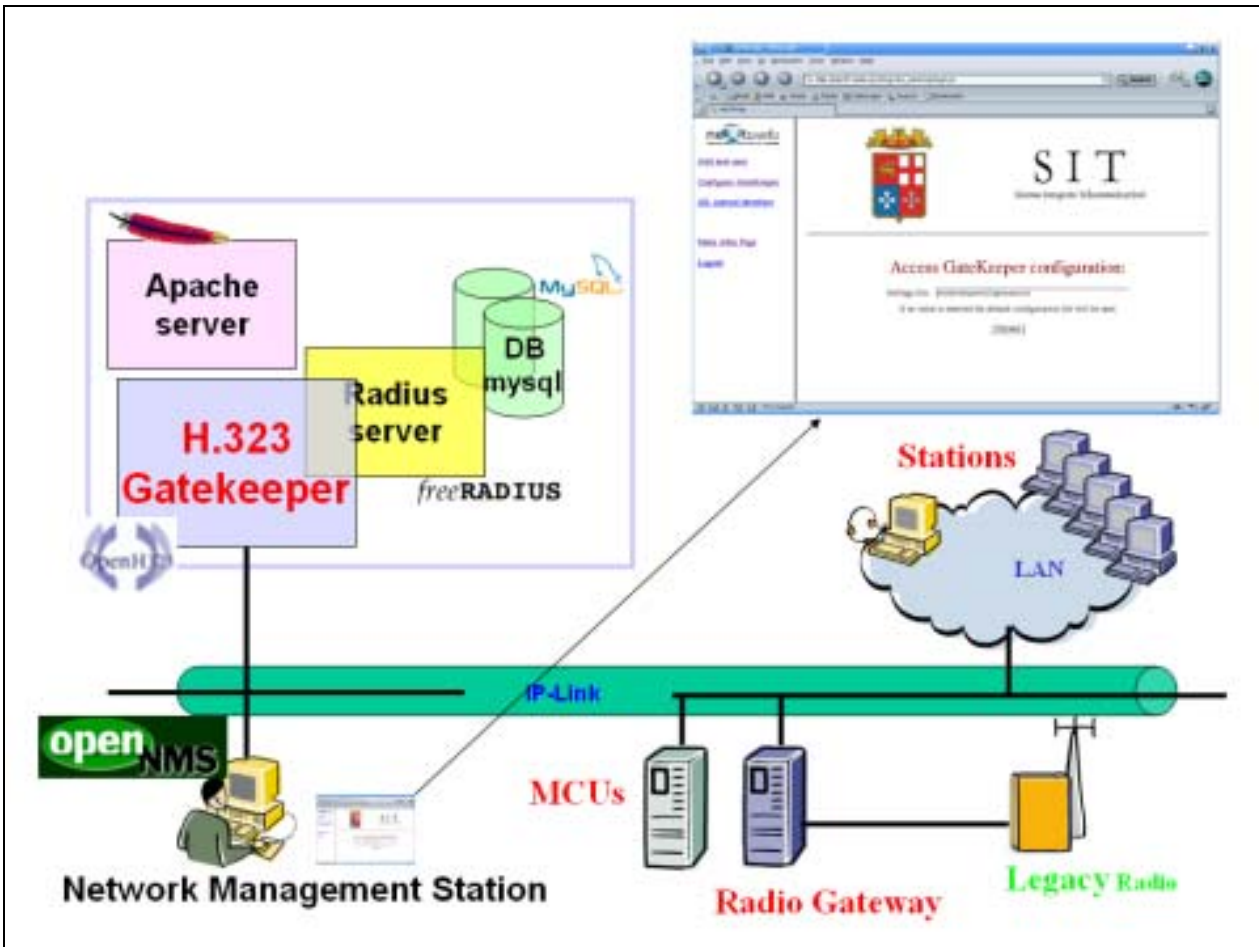
Figure 2: SIT Architecture Overview

The Ext_MCU and the Radio Phone (H.323 softphone) are H.323 terminals based on Open Source applications (open MCU from Openh323 project) modified in order to implement the new SIT services, such as Push to Talk, listing of the users in conference room, channel handling to the radio gateway etc. These terminals are dedicated to the interaction with radio devices not supporting the IP protocol; the communications with such devices is gathered to a H.323 compliant Radio Gateway. In other words, a H.323 gateway is able to interoperate with the legacy radios from a signalling and a data point of view. Figure 3 shows internal signalling and connections that are shared from gateway and radios; the electronic radio enabler that is the signalling transducer that acts as interface between IP and radio sides.
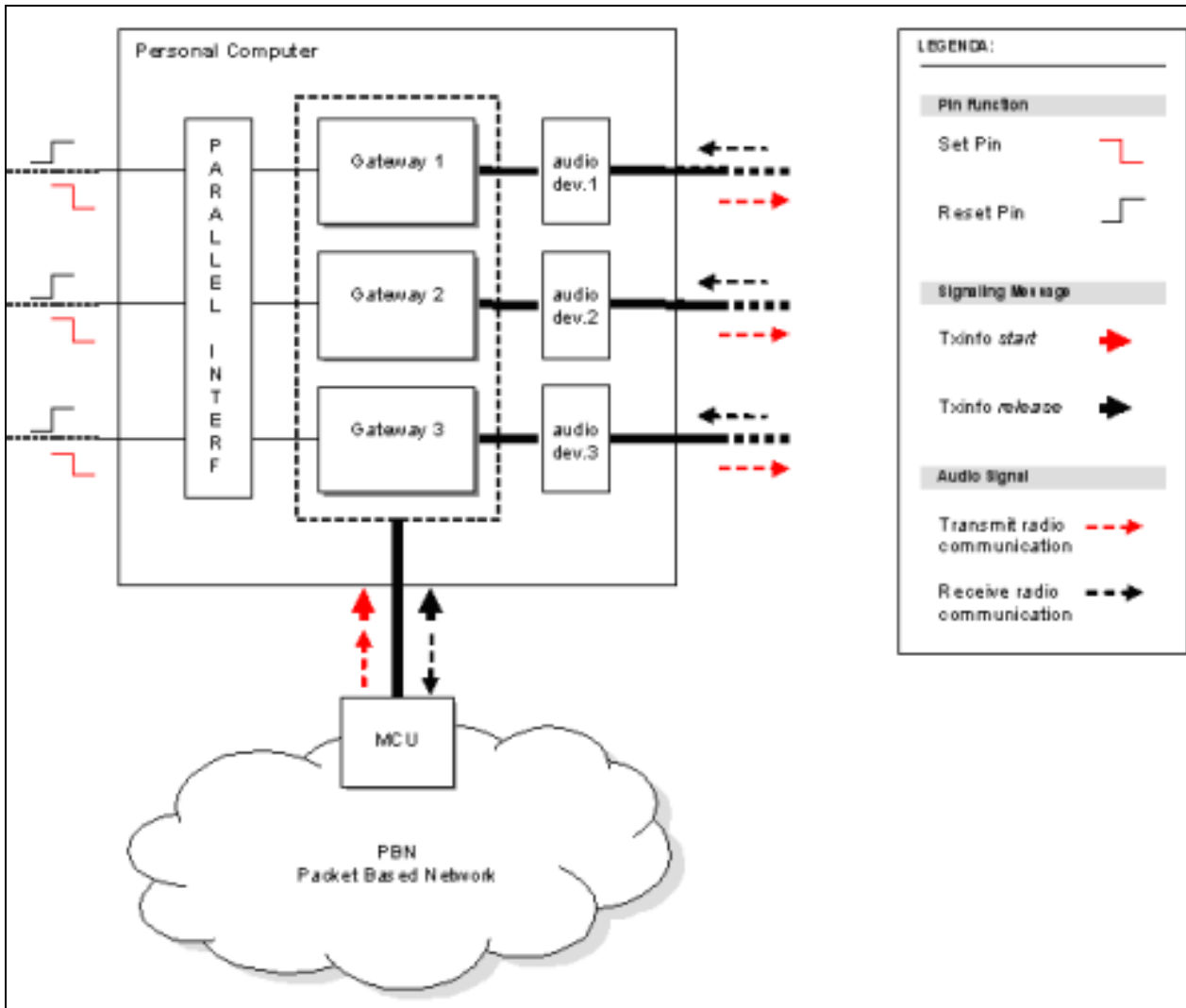
Figure 3: Radio Gateway Signalling

In particular, the H.450 Supplementary Services Framework provides a hierarchical architecture for the creation of new H.323 services. It is based on the client-server architectural paradigm, meaning that endpoints exchange request/response messages. Embedded in these messages are arguments (request messages) and parameters (response messages) that carry the necessary information for the appropriate service. The underlying syntax and structure of these messages relies on X.880, otherwise known as Remote Operations Service (ROS), which defines four operations – INVOKE, REJECT, RETURNRESULT and RETURNERROR. When a served endpoint requires the actions of a supplementary service, it issues an INVOKE message passing it the necessary information in the argument.

Table 1: Description of Messages for new Radio services

Message	From	To	Description
LIST INFO	MCU	Radiophone	Terminals list partecipating in a MCU conference room.
TX RQ	Radiophone	MCU	Transmission request to the radio gateway
TX REL	Radiophone	MCU	Message requesting the release of the channel to the gateway radio
TX INFO	MCU	Radiophone	Message broadcasted to the terminals partecipating to a conference, stating the state of the transmission channel: • *Start* • *Tx on* • *release* This message reports the username sending the request

As two or more entities try to communicate each other, the gatekeeper turns its bandwidth request to the dynamic signalling inter-working unit (described in section 4.3), which interacts with a Bandwidth Broker in the Transport Network. Featuring with dynamic network architectures and bandwidth optimization, this integrated communication system permits direct communications between entities, consequently to the signalling process and the establishment of QoS policies when supported by the core network. Such approach prevents gatekeeper to act as an application proxy and increases robustness of the system.

4.0 THE NETWORK CENTRIC DEMAND FOR QOS PROVISIONING

4.1 SIT Featuring a Dynamic IP DiffServ Network

The applications used in the SIT system intrinsically require an IP network infrastructure able to guarantee the QoS. SIT manages real time services with high interactivity, such as VoIP, together with real time services with a medium-low interactivity, such as videoconference and videostreaming.

The IETF DiffServ QoS architectures [6] are able to provide a framework for Service Providers to differentiate the performance over a range of services that meet different QoS requirements. Although the DiffServ architectures were designed to scale over a large number of users, there still are some weak points when dealing with Service Level Agreements.

Static SLAs are not optimal solutions for users whose demand varies with time. This may often result in periods of insufficient resources available to them or in periods of wasted (but paid) unnecessary resources. In this framework, research activity should be carried out to express how to dynamically manage the network resources. In particular, some relevant issues to address are the definition of appropriate algorithms for Admission Control and for Network Resource Management. A second issue is to find mechanisms permitting the interaction between call signalling (i.e. H.323, SIP) and resource management systems (i.e. Bandwidth Broker, Admission Control Module etc.). Information for these systems are carried out by appropriate signalling protocols (i.e. RSVP, COPS, etc.), defined in the framework of some QoS IP architectures.

The H.323 architecture [7] assigns to the Gatekeeper the tasks of Bandwidth Management and Admission Control. However, in the standard, the algorithms for these two operations are demanded to equipment developers. If VoIP services are supported by a DiffServ network, it is necessary a strict interaction between Gatekeepers and Bandwidth Broker in order to optimise the network resource management, in order to allocate the strictly necessary amount of network resources to guarantee the QoS to all active services.

On the other hand, we need algorithms to set the network elements in order to guarantee the QoS required by the active services. As an example, in [8] the authors analyze two dissimilar strategies to set parameters of a Traffic Control module in a DiffServ network node.

The Traffic Control module implements the traffic metering algorithm. In particular, it is devoted to perform traffic regulation by means of token buckets. Token buckets are simple devices used to constrain the amount of traffic produced by a source to conform a Linear Bounded Arrival Processes (LBAP) description. The volume of traffic A(t) generated by a (λ,B)-LBAP regulated source over any time-interval t is bounded from above by the linear function:

$$A(t) \leq \lambda t + B$$

(1)

For any arbitrary traffic source, there are infinite pairs (λ,B) that satisfy the constraint (1). These pairs lie on a 2D set whose border represents the curve of minimal pairs that satisfy (1) and is referred to as the token bucket curve. The token bucket pair should be selected on the token bucket curve according to some predefined criteria. To select the token bucket pair, in [8] two distinct criteria, namely the knee point and delay bound are compared. The knee point criterion was suggested first by Keshav [9]; it relies upon the empirical observation that token bucket curves generally exhibit a well defined knee area in which their behavior look "regular". Keshav suggests to select the pair (λ,B) within this area; the knee point criterion [10] formally expresses this heuristic as the point with minimum distance with respect to the intersection of the asymptote of the curve with the x axis. The Delay bound criterion was analyzed in literature [11]. It is based on the well known result in [12] by which, in a network of rate-latency schedulers, packets belonging to a (λ,B)-token bucket regulated source experience a maximum end-to-end delay of approximately B/λ provided the source is granted at each node with a capacity λ and a buffer size B. The corresponding pair (λ,B) can be then readily obtained as the intersection of the token bucket curve with the straight line Dmax=B/λ.

The experimental study shown in [8] emphasize the effectiveness of the two different criteria to select the traffic descriptors of VoIP sources, highlighting the higher robustness of the Delay bound criterion. Furthermore, the relevance of the dynamic SLA is experimentally highlighted and the experimental results clearly show the fair nature of the traffic produced by VoIP codecs independently of their characteristics, i.e. CBR or VBR. Indeed, the authors observed the same QoS parameters for the single G.723.1 and G.729 voice sources, when they are aggregated in the same class of service.

4.2 SIT Featuring IP DiffServ Over MPLS Network

The DiffServ network architecture provides basic support to QoS oriented applications. Moreover a considerable effort has been dedicated over the last years to the optimization of operational IP networks: the so called Traffic Engineering [14]. Traffic Engineering is the process of controlling how traffic flows through one's network, in order to optimize resource utilization and network performance. This new feature is needed in the Core Network (i.e. the National Strategic High Speed Network or Internet) mainly because current IGPs (Interior Gateway Protocol) always use the shortest paths to forward traffic. Using shortest paths saves some network resources, but it may also cause two relevant problems.

1. The shortest paths from different sources may overlap on some links, causing congestion on those links.

2. The traffic from a source to a destination exceeds the capacity of the shortest path, while a longer path between these two routers is under-utilized.

In order to perform Traffic Engineering effectively, IETF introduces MPLS [16], which is a forwarding scheme acting between layer 2 (link layer) and layer 3 (network layer) of the OSI reference model.

Each MPLS packet has a short, fixed length (32bits) header, encapsulated between the link layer and the network layer header. A label switched router (LSR) uses the "Label" field of this header to find the next hop as well as the corresponding new "label". The "label" is the encoded Forwarding Equivalence Classes (FEC) to which the packet is assigned. In MPLS network, the assignment of a particular packet to a particular FEC is done just once, as the packet enters the network. When a packet is forwarded to its next hop, the label is sent along with it; that is, the packets are "labeled" before they are forwarded. At subsequent hops, there is no further analysis of the packet network layer header. Rather, the "label" is used as an index into a table which specifies the next hop, and a new "label". The old "label" is replaced with the new "label", and the packet is forwarded to its next hop. In the MPLS forwarding paradigm, once a packet is assigned to a FEC, no further header analysis is done by subsequent routers; all forwarding is driven by the labels.

MPLS requires a protocol to distribute labels necessary to set up Label Switched Paths (LSPs). The Label Distribution Protocol (LDP) and the extended version of RSVP (RSVP-TE) are two different protocols which can be used.

The DiffServ over MPLS model permits to have the advantage of both architectures. This model supposes that a service architecture based on MPLS has been implemented in the core network, but differentiated services must also be supported in order to assure that packets marked with various DSCPs receive the appropriate QoS treatment at each LSR [17]. In this framework, the need for optimal resource utilization and traffic performance entails the adoption of dynamic allocation techniques and admission control mechanisms [18]. To this aim, different approaches have been proposed in literature, but the most interesting is the measurement-based approach. Algorithms based on this approach do not require any a priori knowledge on the traffic, but only traffic data acquired on line during the network operation. As an example in [19], the authors show the design and implementation of a flexible real-time measurement system to be used for management purposes and to be integrated into the network control plane to enable dynamic resource allocation, admission control and traffic engineering on the basis of traffic measurements and predictions.

4.3   A Focus on the dynamic signalling Inter-Working Unit

This section presents a proposal and its implementation for dynamic resource allocation in the SIT environment in DiffServ core network (MOICANE) [20]. We will show that both the access request to the backbone network and the resource reservation are performed by means of a combination of two signalling protocols (H.323 and COPS). The goal of this work is to show that a simple interworking architecture between VoIP and DiffServ can be successfully adopted to provide VoIP users with a scalable and flexible Service Level Agreement scheme. In our proposal, network resources are automatically requested with a combination of the "outsourcing" and the "provisioning" scenario at the call set-up time, avoiding the waste of resources caused by a static SLA definition.
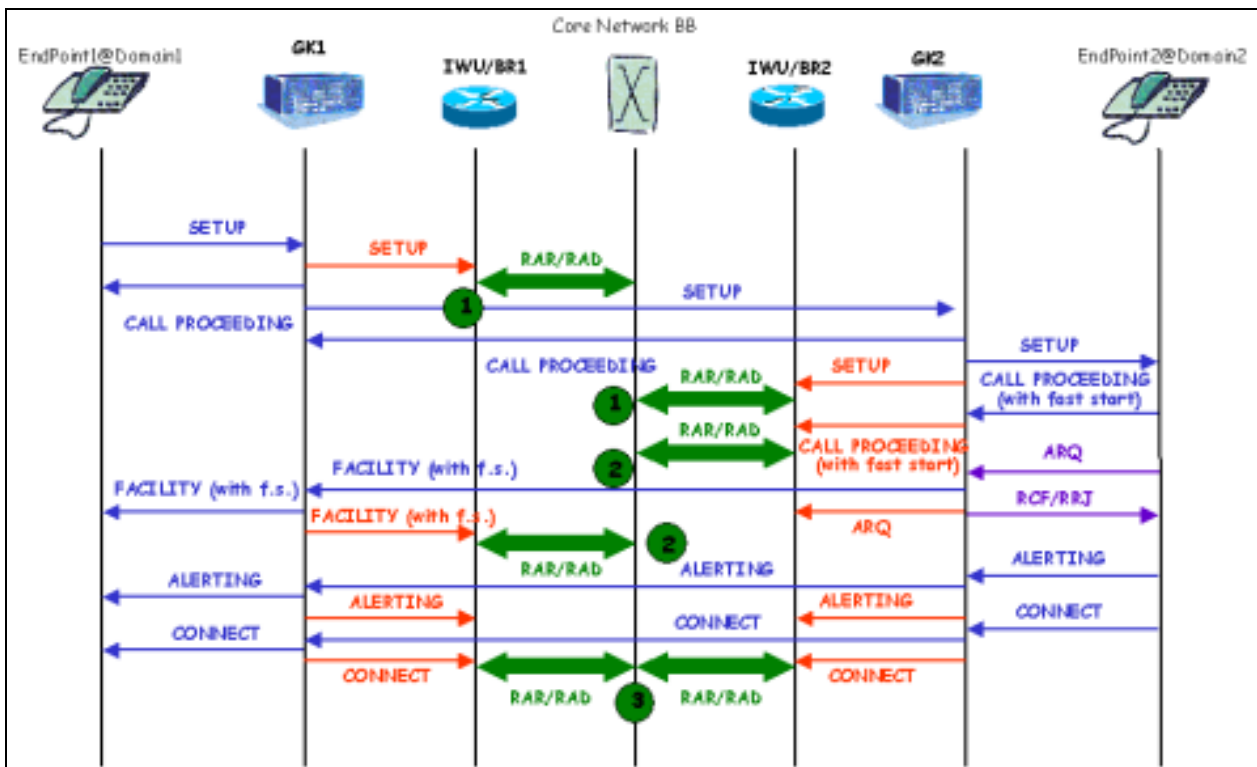
Figure 4: IWU performs IMR and RAR procedures during H.323 signalling exchange

The core element in this scenario, placed at the boundary between the access network and the DiffServ network, is called X-DS BR, where X is a generic architecture (i.e. IntServ, H.323, SIP, DiffServ). The X-DS BR is responsible to understand the signalling coming from the access network, and to accept or refuse the signalling requests according to the resource availability on the DiffServ network and on the established policies. Whatever it might be the access network architecture, the signalling protocol used for authorization or admission control between the X-DS BR and Bandwidth Broker (BB) is assumed to be COPS [21]. So, it is necessary to translate the signalling protocols adopted by each access network into COPS. In order to support multiple access protocol with enough flexibility and easily extend the supported protocols in the future, the architectural choice is to map the different access protocols into a common language, and from this to COPS.

The H.323 Gatekeeper (GK) forwards every H.323 message to the X-DS BR in order to make it locally assemble the COPS messages, which have to be sent to the BB. The X-DS BR has a double role in the control plane: from the H.323 side it has to control the message forwarding from the H.323 GK to itself by means of a simple GKCTRL protocol (with three simple message such as START, STOP and CALL_DELETE), while from the DS side it has to speak with the COPS server inside the BB so requesting resource allocation with an outsourcing request (in the provisioning request the decision is taken locally by the local policy server). The BB is the device in charge of taking outsourcing resource allocation decision communicating with the X-DS BR by means of COPS protocol.

During the H.323 signaling exchange (figure 4, in case the GK is working with at least call signalling routed mode, assuming that fast connet procedure is accepted), the X-DS BR has to perform Incoming Message Request (IMR) and Resource allocation Request (RAR) procedures [22].

5.0   CONCLUSION AND OUTLOOK

In this paper we have shown a proposed design approach able to achieve the challenging features of a modern Integrated Telecommunications System. Describing our prototype SIT we focused on a few characteristics summarizable as follows:

- **Robustly Networked Forces**: SIT represents a services framework enabling communications between nations;

- **Information Sharing plus Dynamics and Flexibility**: With inherent Voice and Video Capability over the TCP- IP suite, SIT exploits the dynamics and flexibility of IP protocol. A simple human interface collects information from different scenarios and media, thus preventing the use of different devices for data and voice;

- **Inclusive and Flexible Acquisition**: SIT is compliant with International Standards and evolves from wide adopted Open Source Projects. Effective interoperability between different producers of standard equipments is therefore matched with full interoperability with COTS and civilian Networks.

Stressing the adoption of Open Standard and Free Software in the design process in order to grant full maintainability of the system and to assure full interoperability between COTS products, we describe our choice of ITU-T H.323 reference framework.

The NEC scenario is composed by a set of networks that can significantly differ (legacy and new technology), having however to be internetworked to grant an effective information superiority. To comply with this imperative feature, SIT integrates a Gateway permitting the full compatibility between standard IP-based communications on one side and legacy Radio Communications System on the other Side, presenting a single human interface.

Considering the kind of services that are required in a real battlefield scenario, we describe the interaction between SIT and DiffServ Networks. Choosing DiffServ Architecture for its scalability, we avoid the use of Static Service Level Agreement (SLA). We propose a way to use Dynamic SLA with traffic control modules in order to optimize network resources. As the simple DiffServ Architecture or the DiffServ over MPLS approach cannot deal with QoS grant services on a per flow basis, as they both use DSCPs for managing routing; we therefore introduce a IntServ over DiffServ approach [23] with the use of H.323 and COPS signalling. The Inter Working Unit collects per flow requests from the H.323 GK and performs Incoming Message Request (IMR) and Resource allocation Request (RAR) procedures in order to ask the requested network resources to the Bandwidth Broker of the Core Network.

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