Preface

Today, mature IP technologies provide an industrial development solution to integrating services into industrial area networks.

This article describes how to create a sound large-scale IP network that combines operational telecontrol and telephony. It highlights the particular measures to be taken at each design phase:

- a topology with an equal balance between path meshing and determinism,
- dimensioning must be based on the exact known flows to be routed,
- complimentarity of QoS mechanisms represents the keystone of IP network performance.

This article discusses the advantages of using QoS mechanisms such as LLQ and CB-WFQ.

The last section deals with the operating conditions for this kind of network.

Keywords: IP (Internet Protocol), service integration, telecontrol, telephony over IP, Quality of Service

INTRODUCTION

RTE (the French power transmission network operator) uses a highly developed Telecontrol Network. Replacing the X25 transmission infrastructure for this network with an IP network also provided the opportunity to integrate an operational telephony service.

Changing to standard technology enables current equipment to be used while adding significant value to components which have become obsolete.

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Equipment maturity alone will not provide the performance level required by demanding applications. The necessary measures still need to be made during the design stage.

After an overview of RTE, this article describes the guidelines for designing a network with a transmission service for telecontrol and telephony, and provides several suggestions for effective operation.

**OVERVIEW**

The French Telecontrol System includes 2200 SS (substations) and more than a hundred high capacity power plants. It also has 76 local and 7 regional control centres and 1 national control centre.

**Telecontrol network top layer**

Applications used by control centres at the top of the network are interconnected by an added-value network which optimises transmission channel costs. This Telecontrol Network (ARTERE), introduced in the mid-90s, aimed to draw on major industrial standards such as Unix, Oracle, and X25 for transmission, a strategy driven by a need for cost efficiency and increased robustness of existing components.

When economically valuable, layered architecture facilitated partial migration to new technologies. In 1996, high level network protocols (OSI) were gradually replaced with TCP/IP protocols while maintaining the existing X25 transmission infrastructure and equipment.

In 1999, the migration to IP was developed further and network interface stations (network stations) were replaced by PCs linked to routers which encapsulated IP in X25.

The announced decline of X25 switches, which today are the backbone of the transmission network, led RTE to look into replacing them with a homogenous IP router network.

**Telecontrol network bottom layer**

RTE also digitised the transmission channels which connect substations, a change that was followed by the implementation of small IP router networks.

**Operational voice**

RTE uses a Security Telephone Network which links operating sites to the control centres. The network is designed to compensate for problems occurring on the public telephone network, in particular congestion.

The renovation of this network due to the obsolescence of certain telephone equipment and the need to reduce analogue connections led RTE to choose a VoIP solution for its new Security Telephone System (STS).

**Service integration**

Almost all the sites concerned by STS are also concerned by ARTERE. Both applications need a simultaneous transmission over IP network.
At the end of 2001, RTE opted for a Multiservice Network (MSN) integrating operational VoIP, telecontrol and remote regulation (cf. figure 1). Teleprotection could not be integrated because of transmission times.

General IP network weak points concerning transmission time and congestion must be considered and dealt with at each network design stage: topology, routing method, dimensioning, quality of differentiated service policy.

*Figure 1: Multiservice network*
DESIGNING A MULTISERVICE NETWORK

Topology
The topology of a network depends on the geographic location of sites. The actual distribution of the RTE Telecontrol Network sites for example, is naturally representative of a tree structure.

The second topological factor to be considered is the meshing level. Increasing mesh size increases robustness, but also connection costs. Conversely, path determinism is reduced, which complicates dimensioning and makes transmission times more uncertain.

RTE has therefore opted for a hierarchic network with a small mesh (cf. figure 2) for the Telecontrol Network top layer, and a limited-sized loop or cluster topology to optimise substation line interconnections.

Adequate availability can thus be obtained by connecting each site using two transmission paths without a common mode.

Routing method
The MSN size is particularly suited to dynamic routing protocol OSPF [1], which can break the network into much smaller areas. The primary area interconnects 7 secondary areas which group together the RCC (Regional Control Centre) and SRCC (Sub-Regional Control Centres) in each region. Substation loops are dealt with as small independent networks.

The main advantage of OSPF is the simplicity of its router setup. The coherence between the addressing scheme and area division limits the size of data exchanged between routers. The network’s small mesh size also enables quick convergence.
Dimensioning
Given that quality of differentiated services alone cannot prevent congestion, network dimensioning is essential in determining real time network transfer capacity.

A network integrating telecontrol and operational voice transmission can be characterised by the significant difference between its average and peak flows. There are few telephone calls during steady-state conditions and ARTERE reduces network congestion from telemetry transmission cycles to a minimum.

This would provide a low minimum flow in the event of a crisis. The random nature of bandwidth requirement cannot gradually regulate flows alone. On the contrary, the network must be dimensioned to dispatch a flow representative of a crisis with telemetry, a predetermined number of voice calls and an avalanche of remote indications.

This last point is particularly important since remote indications must be dispatched in 3 seconds and the volume of the avalanche depends on the electrical incident. The ARTERE protocol is particularly useful in its ability to deal with congestion and to purge old and idle telemetry during exceptional flows. ARTERE thus ensures transmission network efficiency and the flow required can be determined by the maximum size of the avalanche without leading to a purge.

Integrating real time services doesn’t enable the required bandwidth to be reduced, but one 2x transmission channel is still more economical than two x flow channels.

The STS uses the 6.3kbit/sec G723.1 codec with 2 samples per bundle. The cRTP protocol compresses the RTP header, UDP and IP reducing total overhead to just 10 bytes for 48 useful bytes, i.e. a flow of just under 8kbit/sec per telephone call.

Quality of differentiated services
Once the flow is set, QoS (Quality of Service) mechanisms process flows according to bandwidth and dispatch time requirements.

In the MSN, only the voice flow will be classed as being in “real time”. Not only must these packets be dispatched end-to-end in 50 ms, but the variation between packets, or jitter, must not exceed 10 ms.

Telecontrol is more ‘elastic’ in terms of flexibility, since the average bundle dispatch time of 300 ms generally enables the remote indication to be sent to its recipient in 3 s.

Non-priority flows share the remaining bandwidth capacity using a ‘best effort’ type service.

Several protocols need to be placed above the IP in order to reach this level of performance.

Firstly, flows marked by DiffServ mechanisms need to be recognised to differentiate processing. These mechanisms use the DSCP (Differentiated Services Code Point) field coded in the ToS (Type of Service) field of the IP header. This marking enables the LLQ (Low Latency Queuing) to prioritise voice packets.

However, this is not sufficient in an integrated services network. Data packets are around ten times the size of voice packets and their serialisation time on low rate lines can create excessive jitter. They are therefore automatically fragmented by an LFI (Link Fragmentation and Interleaving) mechanism which limits serialisation time to 10ms. The bundle size is thus adapted to the connection flow: e.g. 160 bytes for a 128kbit/s connection.
In addition, to integrate services, part of the bandwidth needs to be limited to high priority traffic to avoid congesting the network and suffocating other flow types.

The CB-WFQ (Class-Based Weighted Fair Queuing) mechanism is used to distribute the remaining bandwidth between the other flows. This mechanism also allows the association of different priorities to various classes while guaranteeing each class a minimal bandwidth on the line.

Each flow can use all the available bandwidth in the absence of other flows. In the event of congestion, each flow keeps the bandwidth defined at the design stage.

Contrary to some protocols in connected mode, where the entire bandwidth is saved for circuit initialisation, here priorities are re-calculated for each bundle and in each network node.

To ensure that the division of bandwidth at the design stage coincides with the actual flow requirements, minimum determinism is preferable when determining flows. There are several ways of proceeding. Firstly a topology with a small mesh ensures that a line, e.g. SRCC-RCC, will pass just one flow originating from two or carry the flow from one SRCC only, rather than two or three at once. Then, a protocol H323 [8], a Gatekeeper and a centralised server enable the Telephone System [7] to ensure that the authorised number of simultaneous calls on a network segment does not exceed the number defined during network design.

**OPERATING THE MSN**

**Reliability**

When integrating a greater number of services, the first issue which arises is that of overall reliability, particularly when operational voice transmission may be used to compensate for a Telecontrol System problem.

Given that separate and integrated networks share the same transmission infrastructure (leased or private lines), service integration can be considered to extend the common mode only at network level, i.e. routers.

In addition, it should not be forgotten that the main purpose of the Security Telephone System is to compensate for problems occurring on the public switched network with which it maintains an interface.

This interface offers several advantages. Above all it facilitates migration while maintaining certain analogue telephone equipment. It also provides a backup for the transmission network, and given the low probability of a simultaneous Security Network failure and public network congestion, still ensures sufficient availability.

As with channels for doubled transmission links without common mode, high network availability is developed at each design stage. In addition to dynamic routing, the MSN uses the HSRP [9] mechanism to automatically select redundant routers from the same site.

All network components provide a x-1 type redundancy. However, they are ineffective without active network monitoring and a suitable organisation to quickly restore component operation so as not to affect applications.
Security
Security is another factor contributing to availability. While implementing widely-used communication standards makes undeniable savings, it also renders the system more susceptible to common technical attacks.

Although the MSN is entirely isolated from external networks and general RTE networks, security has been dealt with at all stages of design. Implementing central network administration with automatic access protection based on TACACS+ (*Terminal Access Controller Access Control System*) and recording all exchanges between routers reduce vulnerability to an acceptable level.

Migration
Migration began mid-2003 and will be completed at the end of 2004.

As illustrated, a ARTERE network architecture, clearly separating the Telecontrol functions from the other TCP, IP and X25 transmission layers, guaranteed a successful and economic migration which did not interrupt operation.

With respect to Telephony, the implementation of gateways managing the H323 protocol at either side of the telephone or PABX analogue lines greatly eases integration with existing equipment.

The Multiservice network can dispatch IP *datagrams* without having to integrate expensive voice interfaces into the equipment. Telephony therefore is integrated like any other application.

CONCLUSION
By describing how to design a large-scale network, this article illustrates that IP technologies brought out around six years ago are mature enough to integrate services, particularly VoIP.

Such technologies are capable of meeting high performance and availability requirements in an industrial environment.

The changeover owes its success to the careful measures taken during design, the ability of existing equipment to migrate progressively, and efficient operational administration.
REFERENCES