Switching and Signalling – The New Architecture.

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Abstract: *This paper presents the new architecture of Switching and Signalling in the telecommunications for the next generation networks. It discusses both the technological and business requirements that have led to the explosion of monolithic switching infrastructure to a decomposed architecture and its components. It briefly describes the functional components and layers of the new architecture and its mapping to the real interfaces and protocols. It is assumed that reader is familiar with both the telecom and IP technologies*.

1. Introduction

At present, we are witnessing a technology shift in the communications industry which involves most avenues of the industry, be it switching, transmission, services, so on and so forth. This paper concentrates on the changes taking place in the switching and signalling architectures of telecommunications: the shift from the TDM to the Packet based technologies (IP/ATM); the explosion of the monolithic switching architectures into decomposed architecture, and the avenues of revenue based on service differentiation.

1.1 The Drivers of change

There are several factors driving the paradigm shift in telecommunications, and in order to understand the new architectures, it is important to understand the reasons behind it. Some of these have been discussed here in brief.

Telephony is no more about a basic call service. It now revolves around customer oriented services. It was realised in mid to late eighties that the network centric and customised services need to be created and deployed with a greater flexibilty and with much shorter deployment times than it was available on the SPC exchanges. The end result was the foundation of the IN architecture which promised all of above, and whilst it delivered most of its promises, it failed to provide the true flexibility and ability to create, e.g. 3rd party services, at much reduced costs. This is an issue which is once again being exploited as one of the basis for the new Switching and Services architecture.

We have seen an unprecedented explosion in the demand for the internet, which has, along with the increased demand for other data services provided by the telecom networks, exposed the inability of rigid telecoms infrastructure to keep up with this demand. The increased data traffic volumes are congesting the core and access networks which were optimised to handle calls with an average of 3 minutes holding time, compared with an average 20 minute holding time for internet sessions.

The global deregulisation also played its role in the opening of the previously closed telecommunications infrastructure, as the new service providers emerged and demanded access to the incumbent's networks and facilities.

And if all of above were not good enough reasons to transform the monolithic infrastructure into a flexible one, the mid nineties saw the introduction of Voice over IP (over internet), which was initially treated as a threat by the service providers, but was later acclaimed by

almost the parties in the industry to be the way forward to fulfill the promise of multimedia, which had failed once before, after the introduction of the ATM and BISDN.
It was all these and many more events that contributed towards a requirement for a flexible infrastructure, which is capable of providing the present and next generation services.
The IP-telephony started with a software based ‘client/server’ internet architecture as a means of cheap long distance calls over the internet, as a toll-bypass. Within no time its conception changed from a hobbyists’ activity to a potential substitute for TDM based telephony, and the standardisation efforts were started by the ITU-T and IETF. If IP-telephony were to gain popularity, it needed to inter-work with the PSTN, and the result was in the from of IP telephony gateways. But the gateways that appeared in the market were by no means capable of providing carrier grade service as expected by the customers, and required by the regulators. These gateway products were merely an enterprise solution with limited performance, scalability, and services – but nevertheless proved a concept, and its far greater potential. The telecom switches on the other hand were comprehensive switching solutions with proven call processing, billing, QoS and other services, which were taken for granted by their customers.
It is at this point in time that Switching and signalling manufacturing industry took a right turn to develop packet telephony system which are underlying transport independent, and began to shape the requirements of flexibility into the new architecture, which will be discussed below. This also led to the efforts to converge the data and telecommunications networks, to create service centric networks, rather than technology dependent network. We are presently witnessing the end results of this: the monolithic switching infrastructure is decomposing, whilst the packet telephony (more precisely packet communications) switches/gateways have begun to gain more functionality. This architecture is supported by the IP telephony standards like SIP (Session Initiation Protocol) and H.323, as well as BICC (Bearer Independent Call Control) and H.248/MEGACO protocols.

2. The new architecture

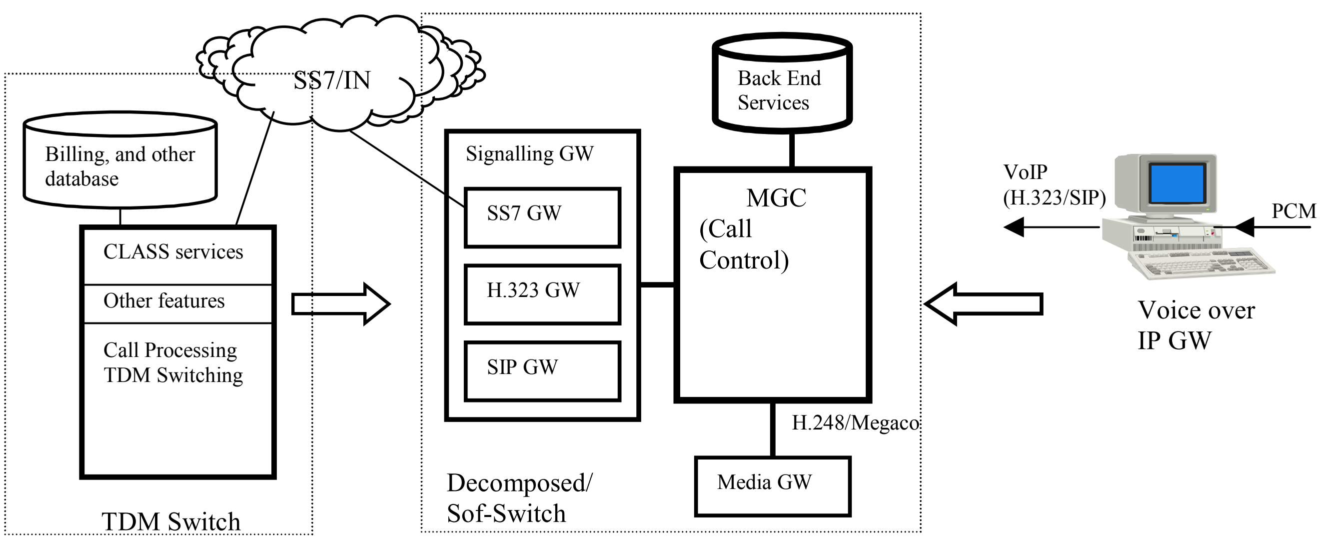


Figure 1 – transformation of a TDM and VoIP switch into a Soft-Switch

The diagram above summarises the changes that have taken place in the switching architecture. It shows the decomposition of a monolithic switch into different components, which will lead to a truly open switching infrastructure. The major components of the above architecture are described below.

2.1 Signalling Gateway.

A signalling GW terminates the signalling to and from the outside world. At present the signalling systems supported are SS7 (with IN), H.323 and SIP - BICC cs2 is under development. Whilst the role of these signalling protocols is outside the scope of this paper, it is important to remember that they are all application layer protocol, utilising IP as transport layer protocols.

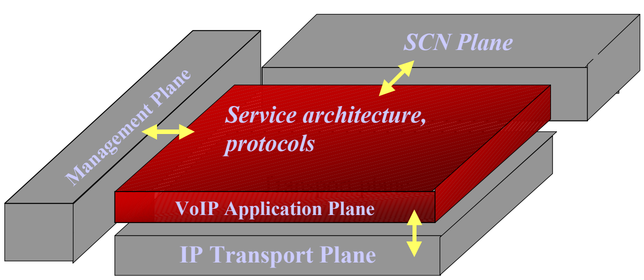
2.2 Media Gateway Controllers

A MGC acts as a controller of the decomposed switch, receiving both the call associated and non call associated requests and acting upon them. It controls the Signalling gateway and the media gateway in order to provide services to its customers. It also performs the tasks of registration, authentication, policy enforcement, as well as provide access to the customised and supplementary services.

2.3 Media Gateway

The MG provides translation from/to different media types, e.g. from PCM voice to packetised voice for transportation over IP (and ATM).
The interface between the MG and MGC is a standardised one, known as H.248/MEGACO, whilst the interface between the Signalling GW and MGC has not been standardised yet.
Note: the above architecture is also known as 'SoftSwitch'.

This architecture can be decomposed further into the planes of functional groupings, which are subject to standardisation efforts by ETSI TIPHON project.



• Fig 2 - Planes

The fig 2 shows the identification and separation of the IP telephony plane from the Management, PSTN (SCN), and IP transport plane. This independence of planes is the basis of the architecture shown in fig 3.

3 The Functional Architecture.

The functional architecture shown in fig 3 has 5 layer corresponding to the IP telephony plane, and not incorporating the transport plane. The architecture for the IP Telephony Application plane consists of functional entities organised into functional layers. One layer builds upon functionality provided by another layer. Together they provide the telephony application. This grouping is useful for the understanding of the functionality involved but does not imply any physical implementation. These functional planes are briefly described below:

3.1 Services Layer:

The services functional layer supports a range of services. These services can be provided either by a service provider of the customer or a 3rd party either locally or remotely. An example of a basic service would be registration. The supplementary services such as Call Forwarding, Call Waiting, CLIP/CLIR would also be deployed at this layer.

3.2 Service Control Layer:

This functional layer controls the services provided to a customer either by the network or the terminal, by providing an interface with the Services functional layer.

3.3 Call Control layer:

This functional layer maintains the context of a call, to provide services such as connection and capabilities requested by a customer, offered by the bearer control functional layer. It is also involved in the transaction of the signalling information with other peer Call Control entities in the same or a different network. In the past, some of the signalling procedures as defined in ISUP were implemented in the Call Control, but the consensus now is to get rid of that model, and have a Call Control Independent of the signalling system used. The Call Control will however support all the current signalling systems including SS7 (Q.931, ISUP, BICC) H.323 and SIP for the Call Control signalling, as well as H.248/Megaco for bearer related signalling, e.g. Media Connection Establishment.

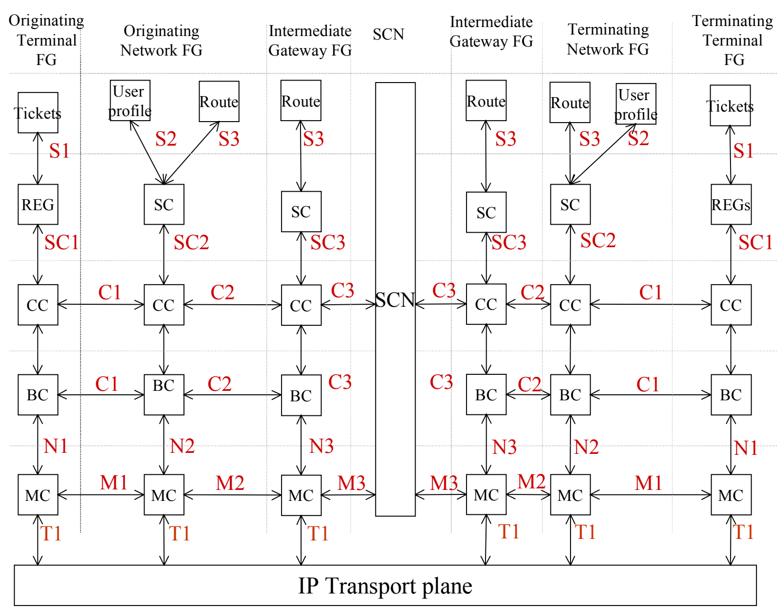


Figure 3 – Functional Grouping of decomposed architecture

3.4 Bearer Control Layer:

This functional layer manages the logical association between endpoints for the media flow, e.g. connect parties A and B.

3.5 Media Control layer

The Media Control layer is responsible for the individual media flows. An example of such functions would be Media encoding alogrithms supported and used, and Firewall control.

The fig 3 shows the ETSI TIPHON scenario 4 [3], where a call originates from an end terminal on the originating network, and is terminated on another network, via several intermediate networks, one or more which could be PSTN or IP telephony networks. There are other scenarios not covered in this paper.

As stated earlier, some of the basic requirements of the new architecture are flexibility, future proof, technology independent, easy service creation and deployment etc. The above architecture provides this with the separation of the bearer, call control, services and the transport planes. One key point to remember, however, is that the above architecture is not

restricted only to voice services, but includes all the multimedia services, where voice is just another, yet an important service.
Another point to remember is that the new architecture does not require a single phase deployment, but can be deployed over time, in different phases, suitable to the customers. A living example of this phased deployment is DLE grooming which is already in operation in several major telecommunications networks across the globe.

There are several reference points shown in the fig 3, details of which can be found in [1]. These reference points are defined by their 'primitives', and 'parameters'. **These reference points are based on the requirements to support communications services and only represent an abstract architecture.** The mapping of these reference points to the real interfaces and protocols is under development. The philosophy behind this architecture is to start with the requirements, as opposed to the protocols and framework available, which has been the case in the recent IP telephony architecture efforts. These requirements are then used to map the real protocols to the required primitives, and find out the shortcomings of the protocol to provide the features and services required.
Consider the example of reference points C and N. 'C' Corresponds to the information flows which are required for the Call Control signalling between peer Call Control entities, employing any of the protocols such as SIP, H.323, BICC, ISUP etc. Whereas, the reference point N corresponds to the information flows between Bearer Control and Media Control functional groupings, to establish, modify and terminate media flows. An example of a real protocol is H.248/MEGACO.
The author of this paper has been involved in the mapping of 'C' reference points to SIP, recording any exceptions and limitations of both the SIP protocol and the above architecture, and taking recommendations to the IETF and ETSI TIPHON for possible changes/additions to the protocol and architecture. The author is also the rapporteur (editor) of the ETSI 'Technical Specification' DTS 3018 in this regard.
The above architecture is not restricted to fixed line services, but is also under consideration in the 3GPP core network [2]. It also brings the two opposite communications models of telephony and internet together, where most of the intelligence in the telecom network is in the core, whilst most of the intelligence in the internet is at the edges. The disadvantage of the former is that the end users have minimal control over their communication services. Whereas, the disadvantage of the latter is that the core networks becomes a transport pipeline providing bandwidth to the applications and services running at the edges. The above architecture liberates the intelligence to the edges while maintaining the service control in the service providers' domain, resulting in a value added service centric network which provides its customers the ability to customise their services.

4 Summary.

This paper provided a summary of the factors affecting the telecoms business, with a focus on switching and signalling. It described the new architecture which is based on the principles of flexibility, future proof, and technology independent. The paper then provided a discussion on the implementation, incorporating the IN.

5 References:

[1] ETSI TIPHON DTS 2003 v9.8. Network Architecture and Reference Configuration.
[2] ETSI TIPHON 19TD52 – TIPHON 3GPP mapping
[3] www.etsi.org/tiphon