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Foreword

To be drafted by ETSI Secretariat.

1 Introduction

This ETSI Technical report investigates the use of ATM as a candidate transport technology in a UMTS Core Network. It outlines the UMTS Core Network Transport requirements based on SMG1 requirements. The second part describes the architectural aspects of a UMTS Core Network based on ATM. This identifies the key interfaces, service and interworking options and outlines the opportunities for ATM usage in the UMTS Core Network. In particular migration from existing networks and co-existence with STM based networks is considered and the QoS mapping between these networks.

2 Scope

This technical report investigates the potential impact of ATM Transport in the UMTS Core Network. It considers the implementation of a UMTS Core Network based on ATM technology including the advantages and drawbacks of the usage of ATM as a UMTS Core Network transport mechanism.

The main part of this technical report describes the benefits and disadvantages in the use of ATM as a potential solution within the UMTS Core Network. This is based on the Transport Requirements in conjunction with the architectural aspects and options described. The report covers the impact of the SMG standard activities in the use of ATM as a UMTS Core Network transport technology with respect to Layer 3 protocols (e.g. GTP).

3 References

3.1 Normative references

This ETR incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references, the latest edition of the publication referred to apply.

UMTS 22.05: "Universal Mobile Telecommunications System (UMTS): Services and Service Capabilities".

Specification of Guaranteed Quality of Service," RFC2212, Sept 1997

Specification of the Controlled-Load Network Element Service," RFC 2211,Sept 1997

Resource ReSerVation Protocol (RSVP)," RFC 2205, Sept 1997

Definition of the Differentiated Service Field (DS Byte) in Ipv4 and Ipv6 Headers," draft-ietf-diffserv-header-00.txt, May 1998.

ITU-T Recommendation I.371 "TRAFFIC CONTROL AND CONGESTION CONTROL IN B-ISDN," May 1996.

ITU-T Recommendation I.356 "B-ISDN ATM LAYER CELL TRANSFER PERFORMANCE," October 1996.

3.2 Informative references

4 Definitions, symbols and abbreviations

4.1 Definitions

Terms introduced in this document:

TBD

4.2 Symbols

For the purposes of the present document, the following symbols apply:

TBD

4.3 Abbreviations

For the purposes of this document, the following abbreviations apply:

AAL-asynchronous transfer mode adaptation layer

ADSL-asymmetric digital subscriber line

ATM-asynchronous transfer mode

BER-bit error rate

BS - Base Station

CBR-constant bit rate

CDMA-code division multiple access

CES-circuit emulation service

CID-connection identification

CPS-common part sublayer

CRC-cyclical redundancy check

DCS-digital cross-connect system

DLCI-data link connection identifier

DTMF-dual-tone multi-frequency signalling

IETF-Internet Engineering Task Force

IP-Internet protocol

ITU-T-International Telecommunication Union - Telecommunication Standardisation Sector

LI-length indicator

LLC-logical link connection

MSC-mobile switching centre

PBX-private branch exchange

PCS-personal communications services

PDC-personal digital cellular

PH-packet handler

POTS-"plain old telephone service"

PPP-point-to-point protocol

PRI-primary rate interface

PSTN-public-switched telephone network

PTI-payload type indicator

SSCF-service-specific convergence function

SSCS-service-specific convergence sublayer

STM-synchronous transfer mode

SVC-shared virtual circuits

UTRAN - UMTS Terrestrial Radio Access Network

VCC-virtual channel connection

VCI-virtual circuit identifier

vocoder-voice coder

VPC-virtual path connection

VPI-virtual path identifier

5 UMTS Core Network Transport Requirements

5.1 Introduction

This section outlines the UMTS CN Transport Requirements including support for:

- Circuit switched traffic (Connection Oriented with guaranteed QoS)
- Packet switch Traffic (Connection Oriented with guaranteed QoS and Connectionless with best effort QoS)
- A wide range of user QoS requirements and traffic profiles
- Real time, non real-time and adaptive flow control services.

These requirements are used as a basis for assessing the use of ATM in the Core Network. The requirements outlined in this document are based on UMTS 22.05: "Universal Mobile Telecommunications System (UMTS): Services and Service Capabilities " being developed in SMG1. In particular the following subsection in 22.05 should be referred to:

- Supported bit rates
- Supported QoS
- Supported topologies and call/session/bearer control features

5.2 Service viewpoint

In the UMTS networks, mobile multimedia services such as voice, data transfer and video services must be provided. The mobile multimedia services, which are assumed to be provided in UMTS system, are shown in Figure 5-1.

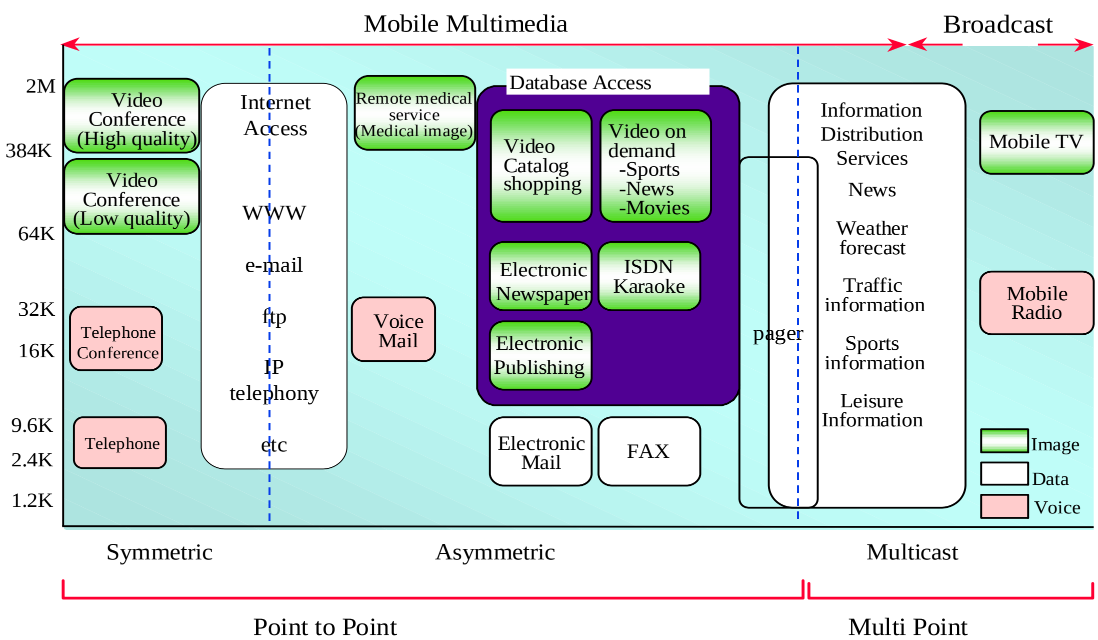


Figure 5-1 Mobile Multimedia Services

These services widely range from the low-speed communication to the high-speed communication up to a maximum of 2 Mbps. A number of communication types are assumed including asymmetrical and symmetrical transmission and Multi- point communication.

The network operator must provide a network environment in which the user can freely use multimedia services without being restricted by the network topologies and the need to re-provision user services. So it is desirable to build an integrated both Circuit Switched and Packet Switched services.

5.3 Summary of Key Requirements

The following requirements are highlighted:

- To enable users to access a wide range of telecommunications services, including many that are today undefined as well as multi-media and high data rates
- To facilitate the provision of a high quality of service (particularly speech quality) similar to that provided by fixed networks
- To provide an efficient means of using network resources
- UMTS service capabilities shall take account of the discontinuous and asymmetric nature of most teleservices and user applications in order to make efficient use of network resources
- The bearer service attributes may be attributed several values when the bearer service required by an application involves more than one connection.
- All the bearer service attributes presented in this clause may be negotiated at call set-up and re-negotiated during the call (mobile or network initiated).
- The UMTS system shall support both connection and connectionless services.
- UMTS shall support four traffic types; constant bit rate, variable bit rate, available bit rate and unspecified bit rate. The UMTS system shall efficiently support variable bit rate services.
- The UMTS system shall allow the efficient statistical multiplexing in a serving network of the traffic resulting from the different mobiles attached to this serving network.

- The delay variation attribute is important for real-time services, e.g. video-conference, where a value approaching 0 would typically be requested.

It is expected that UMTS will be based on a unified transport network. The primary requirements on the unified transport network are:

Support for variable bandwidth capability;

Support for a variety of Quality of Service profiles. This includes the support for :

- Delay sensitive services/applications e.g. voice and video;

- Support for delay tolerant services/applications e.g. e-mail, file transfers;

Support the simultaneous usage of multiple services with different QoS profiles;

Admission Control Functionality;

6 UMTS Core Network Transport Architecture based on ATM

Editor Note: *This section considers the opportunities for ATM usage in the UMTS Core Network. It outlines options within the UMTS Core Network Architecture relating to Transport aspects. This is based on extending the ATM Scenarios developed in UMTS 23.20. The different options identified provide the areas for technical consideration in the evaluation of ATM covered in this section.*

7 UMTS Core Network Transport Control Aspects

Editor Note: *This section considers the UMTS Transport Control aspects. For UMTS it is expected that greater transport flexibility will be required to support functions such as Call Control, Session Control and QoS control. This section outlines these control requirements and describes how ATM might meet these requirements. References are made to the ATM standardisation and timescales.*

7.1 The advantage of the ATM-SVC for QoS support

In the next generation mobile network, it will be desirable to support multimedia services with various required QoS (Quality of service). This section discusses the mechanism to support various QoSs while the network keep high utilisation.

Initially, ATM technology was introduced using ATM-PVCs and is already a major media for the transport of the Internet traffic. ATM-SVC technology is now established and well defined by the ITU-T and the ATM-Forum. It is used not only in many private networks but also for public services.

Using the ATM-SVC technology, bandwidth is allocated on demand, so necessary and sufficient network resources can be assigned for various traffic such as voice, Internet and other data services. Therefore network efficiency is greatly improved.

As with ATM-PVC, ATM-SVC is already used in international network, so it is possible to use ATM-SVC for public network and to support stringent QoS with scalability in 2001/2002.

7.1.1 Necessity of connection oriented approach

There is a requirement form SMG1 that “The UMTS system shall support both connection and connectionless services.” [1]. Switching mechanisms are categorised into two groups, that is, connectionless and connection oriented . A connectionless (CL) switching technique does not require a negotiation phase between user and network cannot control QoS. In the case of connectionless, the QoS depends on the actual traffic to the provisioned network resource.

Therefore, it is basically difficult to control QoS on demand because re-engineering of resources is required to maintain good QoS. In this sense, connectionless is suitable for best-effort type services even though high throughput can be expected since there is no overhead for connection setup. A connection-oriented (CO) approach allows the user to request their QoS and to declare their traffic characteristic, i.e., there can be a call setup phase to establish a traffic contract on demand between the user and the network. In other words, the network can allocate network resources depending on a user's request.

A connection oriented approach with sophisticated admission control can provide various QoS while network resources are used efficiently by statistical multiplexing gain. For example, if the user want a good QoS, the network can allocate the bandwidth based on the declared sustainable rate with a large margin. In contrast, when the user is satisfied with a lower QoS, the network may allocate the bandwidth based on the declared sustained rate with less margin. Of course, if there is insufficient bandwidth to accept a new call for a certain QoS, the network can reject the new call. Admission control needs information such as the requested QoS level and the required source traffic characteristics That is, connection oriented switching technology combined admission control is significant to enhance utilisation of resources to support a variety of QoS.

In the case of an ATM based core network, an ATM-PVC based approach cannot realise on demand based QoS control or efficient resource allocation. On the contrary ATM-SVC technology is suitable to satisfy these requirements.

7.1.2 Mechanism to support QoS

The network must monitor the actual user traffic and prevent the acceptance of access traffic in the data transmission phase in order to guarantee QoS. A policing function is required to maintain the QoS for calls during their holding time. The policing for variable length IP packet is required for IP based network, while the policing for fixed length packet, i.e., ATM cell, is required in the ATM network. The policing for ATM networks can be realised by monitoring traffic in the unit of cell. To realise poling for variable length packets in IP network, a byte count is required. So, it can be said that policing function for ATM is relatively easy compared with that of IP networks. Similarly, the scheduling function for an IP based approach is more complicated than that for ATM.

7.1.3 Standardisation viewpoint

For ATM networks, the traffic contract and ATM Transfer Capabilities (ATCs) such as deterministic bit rate (DBR) and statistical bit-rate (SBR) have been defined in the standardisation organisations (ex. ITU-T, ATM-Forum)[6]. A number of QoS classes with provisioned objective QoS values have been also defined [7]. Further, signalling protocols to carry information elements to support traffic contracts have been specified such as Q.2931 for the user network interface and Q.2761 for the network node interface. ATM-SVC providing various QoS levels can be already developed from the stable recommendations.

Further, to accommodate increasing internet traffic, network architectures to use ATM transport technology for IP packet have been proposed in the ATM Forum and IETF, such as classical IP over ATM and MPOA. These are based on the ATM-SVC scheme when a cut through path for high speed and high quality is established between edge nodes. To establish the ATM-SVC, RSVP can be used. RSVP over ATM technologies are being studied at several organisations such as the IETF. A QoS parameter mapping method between them is expected to be developed in near future. This means that ATM-SVC is recognised in the Internet world.

Within the IETF, Integrated-service [2][3] or int-serv is being developed to provide a guaranteed QoS using a signalling protocol to reserve resources in the router along a path using. The IETF have defined a signalling protocol called RSVP(Resource ReSerVation Protocol)[4]. This was thought to be a promising solution for the Internet by introducing a connection oriented approach, but in practice this is complicated and is not realistic for the Internet router network. It has crucial scalability and billing problems. The IETF are working on a simpler mechanism to deliver QoS with no signalling and with easy metering procedures. Their solution was to simplify control the packet scheduler of each router and gain some sort of differentiated service, or diff-serv(DS)[5]. This aims to provide simple priority control by re-defining DS byte in the IP header. This will not provide an on-demand QoS request. The diff-serv’s approach cannot guarantee QoS provided because it is based on a connectionless approach.

8 System Assumptions

This section describes assumptions necessary for or developing from the evaluation section. It will identify the key system parameters that affect the evaluation.

The following assumptions are taken from SMG12 23.30:

1. Transport protocol across the Iu interface for UTRAN shall be based on ATM.
2. The Iu shall support all service capabilities offered to UMTS users
3. Iu shall particularly cater for a variety of services e.g. classical telephony, internet-based services (www, e-mail etc.), and multimedia services. This implies that Iu supports efficiently:
    - dedicated circuits, especially for voice
    - best-effort packet services (e.g. Internet/IP)
    - Real-time multimedia services requiring a higher degree of QoS. These real time services may be based on real-time packet data or circuit-switched data.
4. UMTS Phase 1 (Release 99) network architecture and standards shall allow the operator to choose between Integrated and Separated core networks for transmission (including L2).

9 Evaluation of ATM in a UMTS Core Network Transport

Editor Note: *This section considers how ATM might meet the UMTS transport requirements. It describes the advantages and drawbacks of using ATM as an integrated UMTS Core Network transport mechanism on the Iu-interface and inside the core network domain. It considers the different architectural options and evaluates the requirements for these. References are made where appropriate to simulation and modelling activities to support the work.*

9.1 The efficiency of ATM and IP Network

This section will compare ATM network and IP network from the view point of network efficiency. For data traffic statistical multiplexing gains in both networks are evaluated. For voice traffic, transmission efficiency in both ATM with AAL2 network and UDP/IP network are evaluated.

9.1.1 Data traffic

This section compares the required buffer sizes in a core network composed of cell switches (ATM) and a core network composed of packet switches (IP network) using queuing theory.

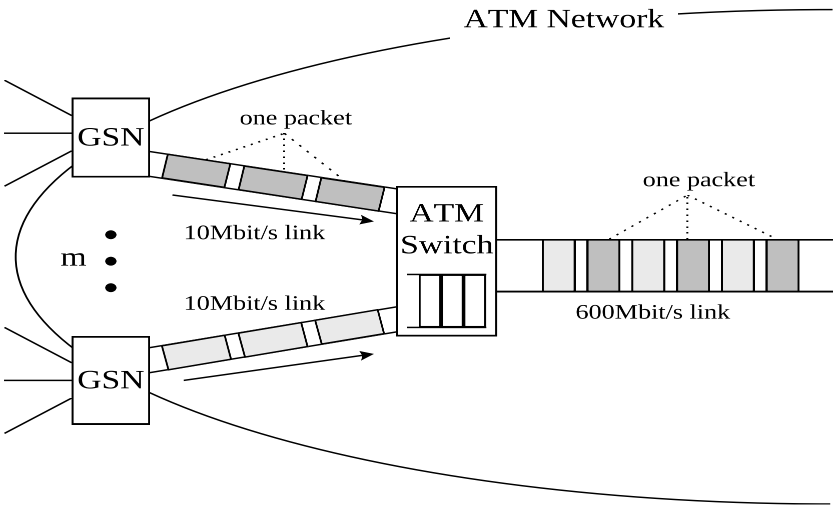
Figure 9-1 shows the structures of a cell switching network (shown as (a)) and a packet switching network (shown as (b)).

The following network and traffic models are assumed;

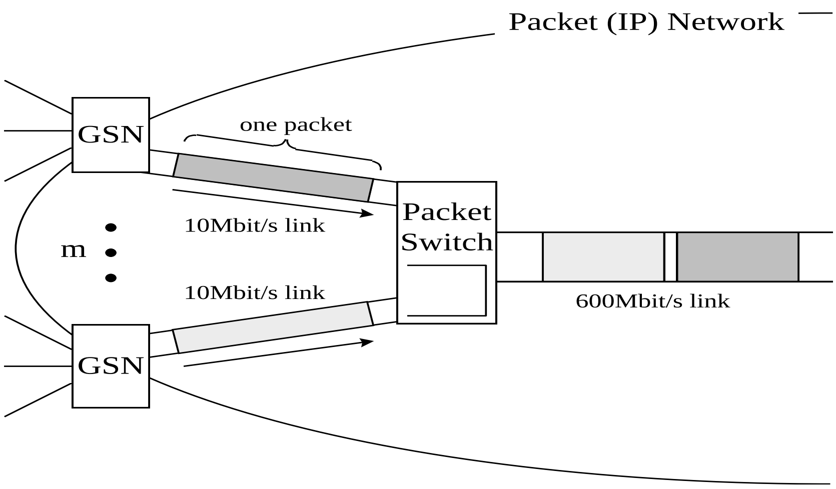
- Peak rate (1 traffic source): 10Mbit/s (the link bandwidth between GSN and switch),
- Sustainable rate (1 traffic source): 1.5 Mbit/s,
- Average packet size: 250 bytes (exponential distribution),
- Backbone link bandwidth (between switches): 600 Mbit/s,
- Used queuing model:
  - MMPP/D/1 (cell switching),
  - M/M/1 (packet switching),
- Traffic volume: variable,
- Target loss ratio,
  - 1×10^{-7} (cell loss),
  - 5×10^{-7} (packet loss).

Using traffic model for cell switching is shown in Figure 9-2.

The required buffer sizes for backbone links of both switches were compared. For the cell switching, the required buffer size is the product of the number of cells queued and the cell sizes (53 bytes). For packet switching, the buffer size is the product of the number of packets queued and the average packet sizes (250 bytes).



(a) Cell Multiplexing on ATM Network



(b) Packet Multiplexing (on IP network)

Figure 9-1 Network Model

Figure 9-3 shows the results of the analysis. As shown, the required buffer size of cell multiplexing using ATM switches is smaller than that of the packet multiplexing using IP switches. In other words, if the same buffer size is used in both networks, the maximum link utilisation of the ATM network is larger than the IP network. Therefore the cost of transmission systems in the ATM network is lower than in the IP network.

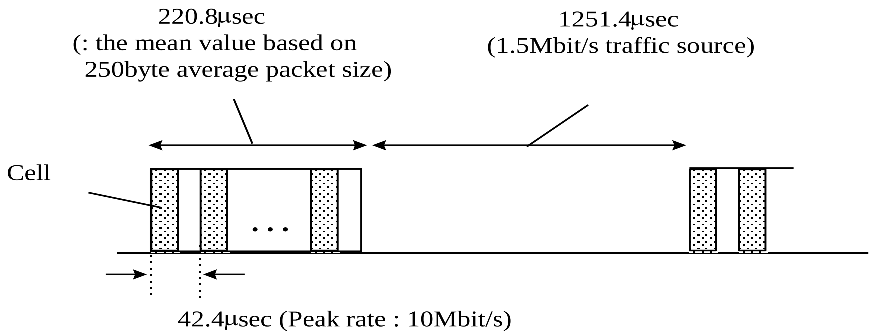


Figure 9-2 Burst traffic model for cell switching

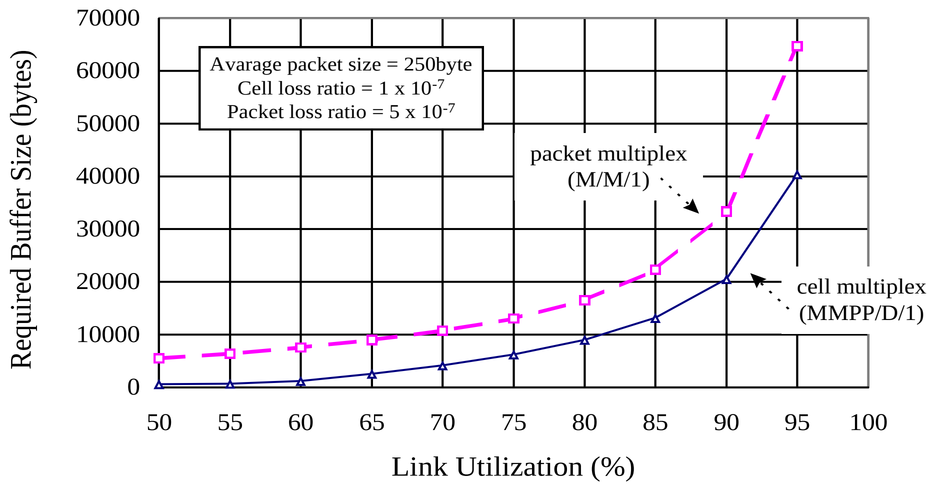


Figure 9-3 Comparison of Required Buffer Size

9.1.2 Voice Traffic

This section describes the estimations of the simultaneous connections over the physical layer interface and the transfer overhead using ATM (especially AAL-2) technology for voice data transfer.

1) Link Efficiency

Voice data is encoded by a specific encoding algorithm and transferred over the radio interface in order to save radio resources. It is possible to save more resources if a silence suppressing mechanism is used. ATM, especially AAL type2, is effective to transfer voice data over CN. Figure 9-4 shows the simultaneous connections over the physical layer interface.

Note: AAL2 short packet length ： 13 octets／ 10ms (10.4kbit/s) (ITU-T G.729)

Link utilisation of an interface: 80%

Talk-spurt rate: 50%

For example, using a 50 Mbit/s STM interface, about 800 channels can be transferred. If a voice coding method, such as ITU-T G.729, and AAL type2 technology are used, more than 3,800 channels can be transferred over the same interface. Moreover, up to 7,700 channels can be transferred over the interface using a silence suppression mechanism.

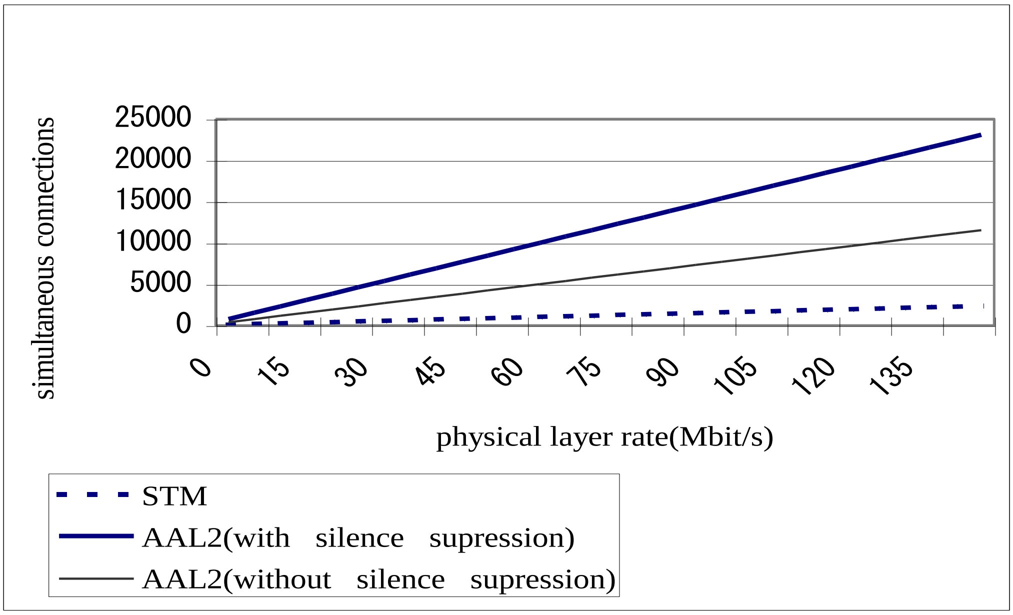


Figure 9-4: Simultaneous connections using AAL type 2

2) Transfer Overhead

A speech data unit is encapsulated according to the lower layer segmentation or packetization method. Address and control information are added to the data unit for network routing purposes. For example,

UDP/ IP---UDP header (8octets) and IP header (20octets) are added to each data unit.

AAL2---short packet header (3octets) is added to the data unit, if the data length is shorter than 45 octets. And an ATM header is added (per 47 octets)

Figure 9-5 shows the relationship between the data length and the packet length (total transfer octets).

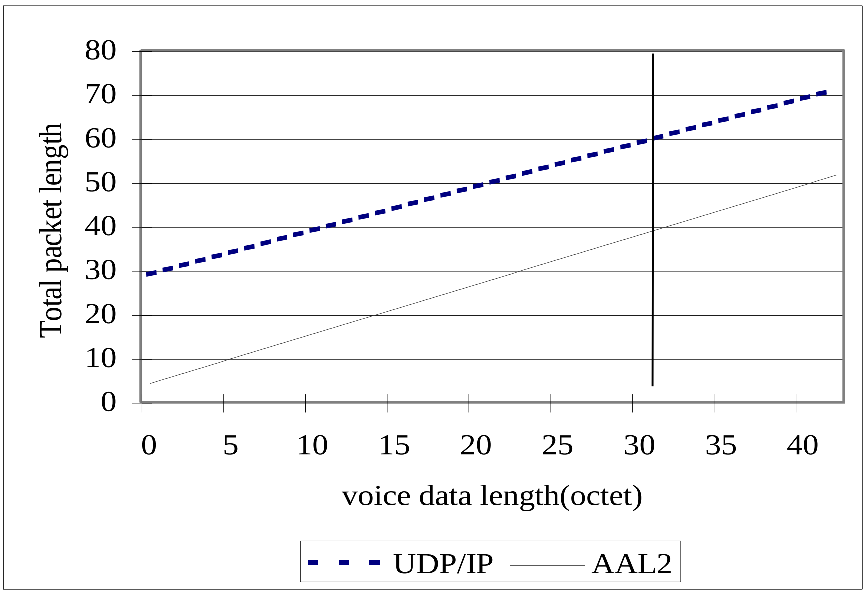


Figure 9-5: Voice data length and Total packet length

When the encoding process is performed on a 20 ms speech frame and the speech data rate is 12.8Kbit/s (then the data length is 32 octets), the UDP/IP packet length is 60 octets. On the other hand, the data unit can be transferred in a single AAL type2 short packet using AAL type2 packetization. Then total transfer octets are less than 40. When the data length is short such as voice data, the transfer overhead of AAL type 2 is smaller than that of UDP/IP. So AAL type 2 is suitable for voice data transfer.

9.1.3 Conclusion

For the data traffic, the statistical multiplexing gain of the ATM is lager than that of IP packet network. For the voice traffic, the usage of ATM with AAL type 2 can provide the capability to much more voice connections than the usage of UDP/IP. Therefore, ATM transport technology is suitable solution from the viewpoint of efficient network usage rather than IP based solution. Further, to adapt the various user requirement on QoS with efficient network resource, traffic control such as admission control based on negotiation by signalling are necessary. Further, many actual ATM-SVC based QoS control have been developed. Therefore, this contribution concludes that ATM-SVC is the most appropriate technology to support various QoS requirements and to use network resource efficiently. IMT-2000/UMTS Core Network (CN) phase 1 should include ATM-SVC capability as one potential and realistic solution.

9.2 Analysis of Bandwidth Efficiency, Delay and Jitter

This section analyses the bandwidth efficiency, delay, and jitter associated with AAL-2 multiplexing. While the analysis has been carried out for many of the applications discussed earlier, we are presenting here the results for two specific wireless applications for an alternative point of view: the IS-95 CDMA rate set 2 vocoder 6 and the Japanese personal digital cellular (PDC) half-rate vocoder 7. The conditions and results apply equally well to the E1 and E3 with GSM case, with which we are more familiar. Performance of AAL-2-based transport for the above two applications is compared with performance when using frame-relay and STM transport.

9.2.1 Traffic and Models

To determine the bandwidth efficiency and delay/jitter trade-offs, specific traffic patterns and speech models are used. It is assumed that only voice traffic is carried on the ATM connections of interest. In practice, some in-band signalling and OA&M traffic may exist but their overall impact will be slight.

9.2.1.1 CDMA rate set 2 vocoder

Various levels of coding have been defined for the CDMA rate set 2 with a vocoder full rate of 13 kb/s. A simplified speech model is used here for the studies. Specifically, 50% of the packets are at the full encoding rate of 14.4 kb/s (active speech, including overhead), while the remaining 50% are at the one-eighth rate of 1.8 kb/s (silence). The overall average rate is 8.1 kb/s. This coder produces a measured mean opinion score better than 3.95. Speech is accumulated for 20 ms, encoded, and transmitted over the air interface as a packet.

9.2.1.2 PDC half-rate vocoder

The same simplified speech activity model for the PDC half-rate vocoder produces an average rate of 2 kb/s with a packet produced every 40 ms at either the full rate (4 kb/s) or complete silence (0 kb/s) with equal probability. This vocoder provides an extreme point with a very low bit rate resulting in maximum bandwidth efficiency.

9.2.2 Requirements on Delay Variation

In IS-95 based CDMA technology, voice packets are produced every 20 ms. Because of soft handoff, the reception of packets at the mobile must be synchronised perfectly. In addition, because of tight timing requirements, all base stations must be synchronised using the Global Positioning System.

It is possible for all the mobiles to transmit packets at every 20-ms tick. In this case, the delay variation can be up to 20 ms depending on the number of active calls. For interactive voice applications, experiments have determined that the one-way end-to-end delay must be in the range of 100 to 150 ms. Because coding, interleaving, decoding, and de-interleaving can consume a substantial portion of this delay budget, an additional delay of 20 ms is on the boundary of acceptable performance.

The transmission of packets from mobiles can be staggered such that one set of mobiles transmits at a given time tick, a second set transmits exactly 5 ms later, a third set 10 ms later, and the last set 15 ms later. These are referred to as offset groups. Two cases will be considered: one in which the requirement on the maximum delay variation is 5 ms (that is, four different offset groups) and the second in which the maximum permissible delay variation is 20 ms (that is, one offset group).

9.2.3 Results and Discussion

First, consider the CDMA rate set 2, which produces packets containing 36 octets and 5 octets with equal probability. The AAL-2 adds an overhead of 3 octets to each packet. The effective ATM cell payload is 47 octets because the first octet is used as the STF in every cell. In the case of frame relay, there is an overhead of 6 octets for each packet. (The Frame Relay Forum is currently standardising an approach similar to the AAL-2 to carry multiple small packets within one data link connection identifier (DLCI). The efficiency gains from this are not considered here). Frame-relay overhead consists of one octet for flag, two octets for the DLCI field, one octet for control, and two octets for the frame check sequence.

Frame relay, with its variable-size frame, is ideally suited for carrying variable-size packets generated by low bit rate voice. The AAL-2 provides a similar capability over ATM connections. It also allows the use of much higher speed ATM switches and link interfaces, thus allowing further multiplexing gain. Finally, with higher speed interfaces, ATM transport and switching are much less expensive than the frame-relay counterparts. If ATM can achieve bandwidth efficiency comparable to that of frame relay, lower switching cost and the ability to support higher rate interfaces will favor ATM.

Transmission facility (Mb/s)	Maximum Delay Variation (ms)	Number of Voice Calls Supported			
		AAL-2	Frame Relay	TDM	AAL-1/AAL-5
T1 (1.536)	20	123	125	24	72
T1 (1.536)	5	104	108	24	72
T3 (44.7)	20	4,090	3,500	672	2108
T3 (44.7)	5	3,964	3,024	672	2108

Table 9-1: Number of Voice calls supported for CDMA rate set 2

Table 9-1 shows the number of voice calls that can be transported by the AAL-5, the AAL-2, Frame Relay, and STM transport as a function of transmission speed and the maximum allowable delay variation. For STM, each voice call uses a 64-kb/s channel out of a T1 or T3 interface. For AAL-1/AAL-5, it is assumed that one voice packet is carried per cell.

At T1 rates, both frame relay and the AAL-2 are equally efficient. At T3 rates, it is possible to achieve greater gains via statistical multiplexing using the AAL-2. The difference in call carrying capacity is as much as 30% between frame relay and the AAL-2 when the overall demand exceeds the frame-relay interface speed (T1).

Another interesting point is that the difference in call carrying capacity between delay variation objectives of 20 ms and 5 ms is 18% at T1 speeds, while this difference is less than

4% at T3 speeds. Even if the T3 rate is divided into four virtual circuits, each having bandwidth equal to one-fourth that of T3 capacity, the resulting bandwidth is sufficient to attain statistical multiplexing gain. Thereafter, the improvement in this gain is marginal.

From the call carrying capacity values at T1 and T3 speeds, we can see that it is better to carry voice calls in ATM connections of larger bandwidth rather than partitioning the available bandwidth into multiple CBR virtual connections. Because a CID field of 8 bits limits the number of LLCs for a given ATM VCC to fewer than 256, further statistical gain can be achieved by implementing higher rate VPCs (for instance, T3 or fractional T3 VPCs). Such gain can be realised by using multiple VBR VCCs within the VPC and then using the 8-bit CID field to specify individual LLCs. Policing at the VPC level makes both the VCI and CID fields available to the end points for addressing while achieving multiplexing gain corresponding to the VPC speed.

The call carrying capacity of STM transport, wherein each voice call is mapped into 64 kb/s PCM voice at the base station, shows the advantages of asynchronous transport (frame relay and ATM) over synchronous transport. It is quite apparent that the call carrying capacity is increased 500% if such statistical multiplexing techniques as ATM or frame relay are used for voice transport.

The bandwidth efficiency achievable using AAL-1 or AAL-5 is also shown in Table 9-1. The call carrying capacity achieved by the AAL-2 is 1.8 to 2 times that of AAL-5 or AAL-1.

Transmission facility (Mb/s)	Maximum Delay Variation (ms)	Number of Voice Calls Supported			
		AAL-2	Frame Relay	TDM	AAL-1/AAL-5
T1 (1.536)	20	500	500	24	224
T1 (1.536)	5	420	420	24	176
T3 (44.7)	20	16,680	14,000	672	8,050
T3 (44.7)	5	16,160	11,760	672	7,688

Table 9-2: Number of Voice calls supported for PDC half rate

Table 9-2 shows the call carrying capacity for PDC half rate. In general, the observations for CDMA calls also hold for PDC half-rate calls. However, on facilities in which the maximum allowable delay variation is 5 ms, the additional gain in statistical multiplexing achievable by the AAL-2 is 28% compared to frame relay. When the maximum delay variation is 20 ms, the additional gain in statistical multiplexing is 17%. Note that clever use of VPC, VCC, and CIDs will again be needed to achieve the full statistical multiplexing offered by high-speed ATM interfaces.

There should be multiplexing at the Iu interface to take advantage of higher speed ATM interfaces in the case that the individual Base Stations do not handle very high traffic. By mapping multiple ATM VCCs from individual BSs into one VPC between the BSC and PH, high bandwidth efficiency can be achieved even with low CID size.

9.2.4 Rebundling

The AAL-2 efficiently transports short variable-length packets over an ATM connection spanning many ATM switches. As mentioned earlier, termination points of AAL-2 connections may be at Base Stations and MSCs in wireless cellular/PCS applications, as well as PBXs and voice gateway severs in other packet telephony applications.

Current standards define point-to-point AAL-2 connections (LLCs) over an ATM VPC or VCC. Many applications may have connections originating at one point and ending at many different destinations. In these cases, creating multiple ATM VPCs or VCCs such that each can serve as a vehicle for a subset of point-to-point AAL-2 connections may lead to a small number of LLCs per ATM connection and, hence, to a significant loss of efficiency. An example is given below.

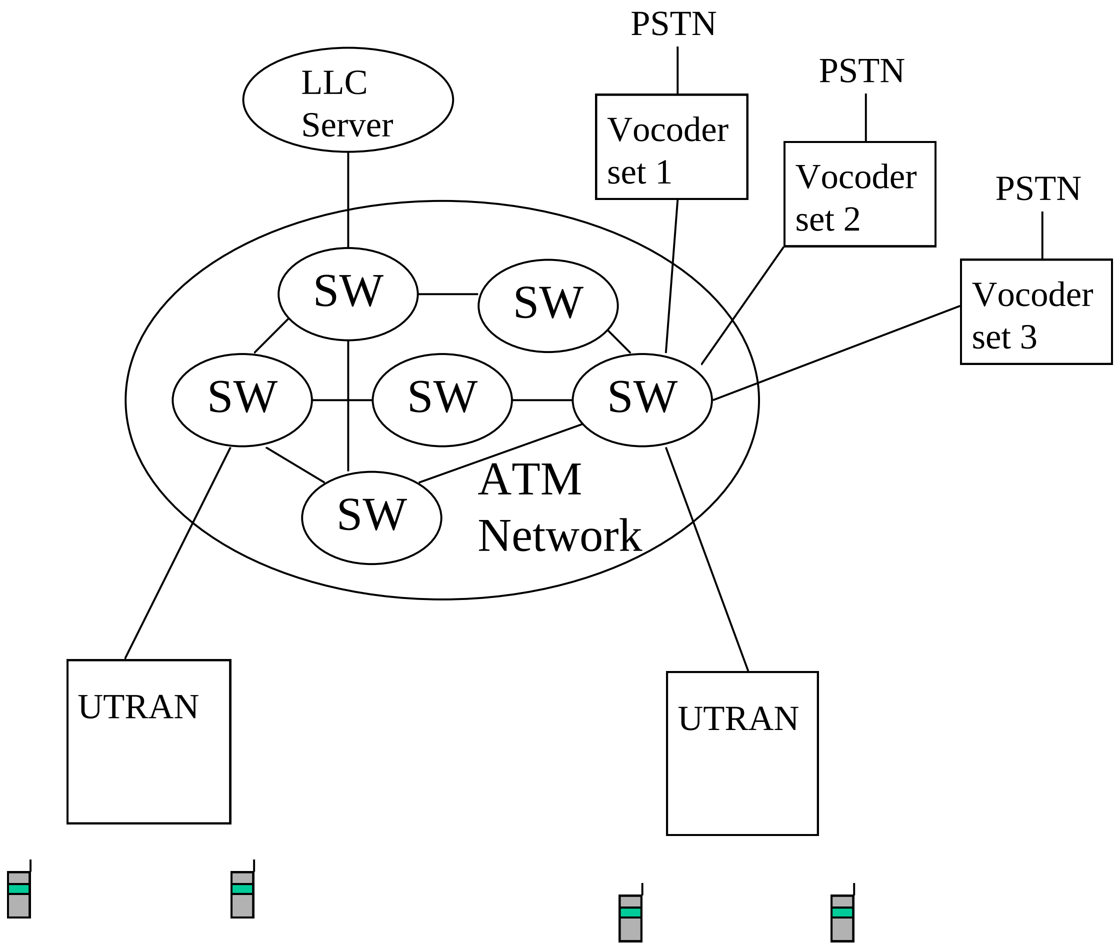


Figure 9-6: The Role of the LLC Server in Improving Bandwidth Efficiency

Figure 9-6 shows one such example, the ATM cloud could extend down into the UTRAN. While an ATM switch may handle a large number of Base Stations, connections at BSs may use different PSTN interfaces. Typically, the association between a BS and the PSTN interface is dynamic due to mobile hand-off. At hand-off, the BS changes but the PSTN interface remains anchored. Thus, an ATM connection between an BS and a vocoder set may not have enough LLCs to achieve high multiplexing gain.

One suggested solution to this problem is to have two ATM connections in the path of an AAL-2 connection, one between the BS and LLC server and another between the LLC server and the vocoder set. All LLCs from and to a given BS use a common ATM VPC or VCC irrespective of the vocoder set at the other end. Similarly, all LLCs from and to a given vocoder set use a common ATM VPC or VCC irrespective of the BS at the other end. At the LLC server, LLC packets from an BS are extracted and multiplexed into the ATM connection between the LLC server and the particular vocoder set. A similar procedure applies to the packets originating at vocoder sets and destined for BSs.

In fact, the situation is similar to that of the traditional core transport network consisting of different types of digital cross-connect systems (DCSs), line termination units, and fiber routes. DCSs and LTEs act as rebundling devices for lower rate circuits over very high capacity fiber routes. Thus, for broader applications of the AAL-2 with multiple end points, LLC servers may be desirable at many places in the network.

In practise the right balance between PVCs and SVCs in the Radio Access Network is required according to the differing needs e.g. an SVC will be the best approach to support transaction orientated communication (connectionless), whereas, PVC will be used to carry signalling.

9.2.5 Summary

The new AAL-2 has several applications. In particular, it achieves high bandwidth efficiency and low packetisation delay simultaneously, thus making it ideal for voice and other low bit-rate interactive applications. While the basic common part of AAL-2 is standardised, signalling requirements and alternative transport mechanisms for signalling messages are the focus of current studies e.g. AAL-5 for high speed data (internet).

Further work on LLC servers is in progress. Specifically, the locations of LLC servers and routing of LLCs using one or more LLC servers is being worked as a network design problem.

The basic principles of the AAL-2 are also applicable in non-ATM environments. Both the Frame Relay Forum and the IETF are investigating multiplexing protocols similar to the AAL-2 for carrying low-bit-rate voice over frame-relay and IP networks, respectively.

10 Impacts on UMTS and Recommendations

Editor Note: *This section describes the recommendations and conclusions on the use of ATM in the UMTS Core Network. This includes the impact on the SMG standardisation activities in the use of ATM as a UMTS Core Network transport mechanism.*

10.1 GTP

Assuming GTP as tunnelling protocol, both of SGSN (or GGSN) in the ATM SVC tunnelling network and GGSN(or SGSN) in the IP tunnelling network are necessary to be connected by GTP tunnelling. Regarding the connection with 2nd GPRS node and IP based GPRS node, connection IP base among all GSN nodes address in the IMT-2000/UMTS network should be IP address. In this context, GTP should be carried with IP address through IP base transfer mechanism.

Figure 10-1 shows the U-plane protocol stack. Appendix A shows the sequences for mobile originating, mobile terminating and handover.

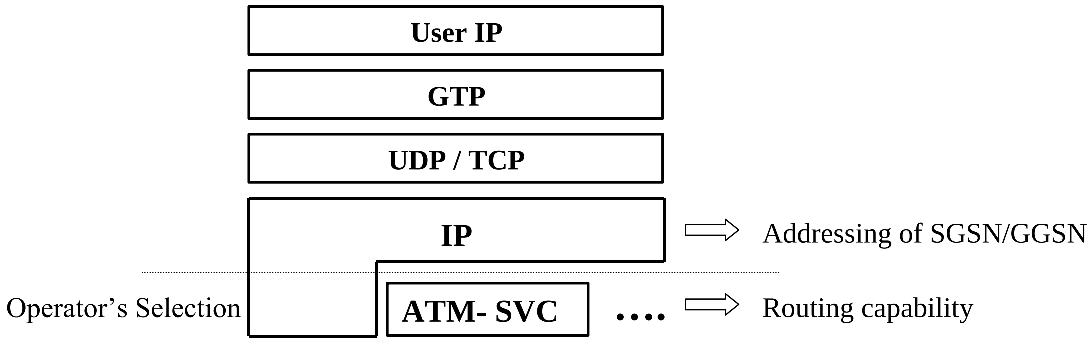


Figure 10-1: U-Plane Protocol Structure for Packet CN

In UMTS/GPRS, it should be possible for operators to use different packet switching protocol (e.g. ATM-SVC) under single GTP standard.

Between GSNs, GTP uses UDP/IP (or TCP/IP) for addressing regardless whether IP routing or ATM-SVC switching is used. The use of ATM-SVC will not impact on GTP standardisation.

All GSN node address of GTP layer are IP addresses.

There is no need for IWF between IP based CN and ATM-SVC based CN.

There is no node other than SGSN and GGSN which perform adding/deleting/modifying any parameters in the GTP messages. Transport layer converter may refer to the relating GTP messages only to manage/control of ATM SVC but does not have any funciton for changing parameters in the GTP messages.

Only transport layer converter (IP addressing to/from ATM address) is necessary for the interworking between the UMTS networks.

Converting IP address in GTP to ATM address will be performed in the transport layer converter.

10.2 Others

11 References to ETSs

12 Appendix A

The following patterns are shown in this appendix.

(A) Originating Call

(B) Terminating Call

(C) Handover between SGSNs (Same CV)

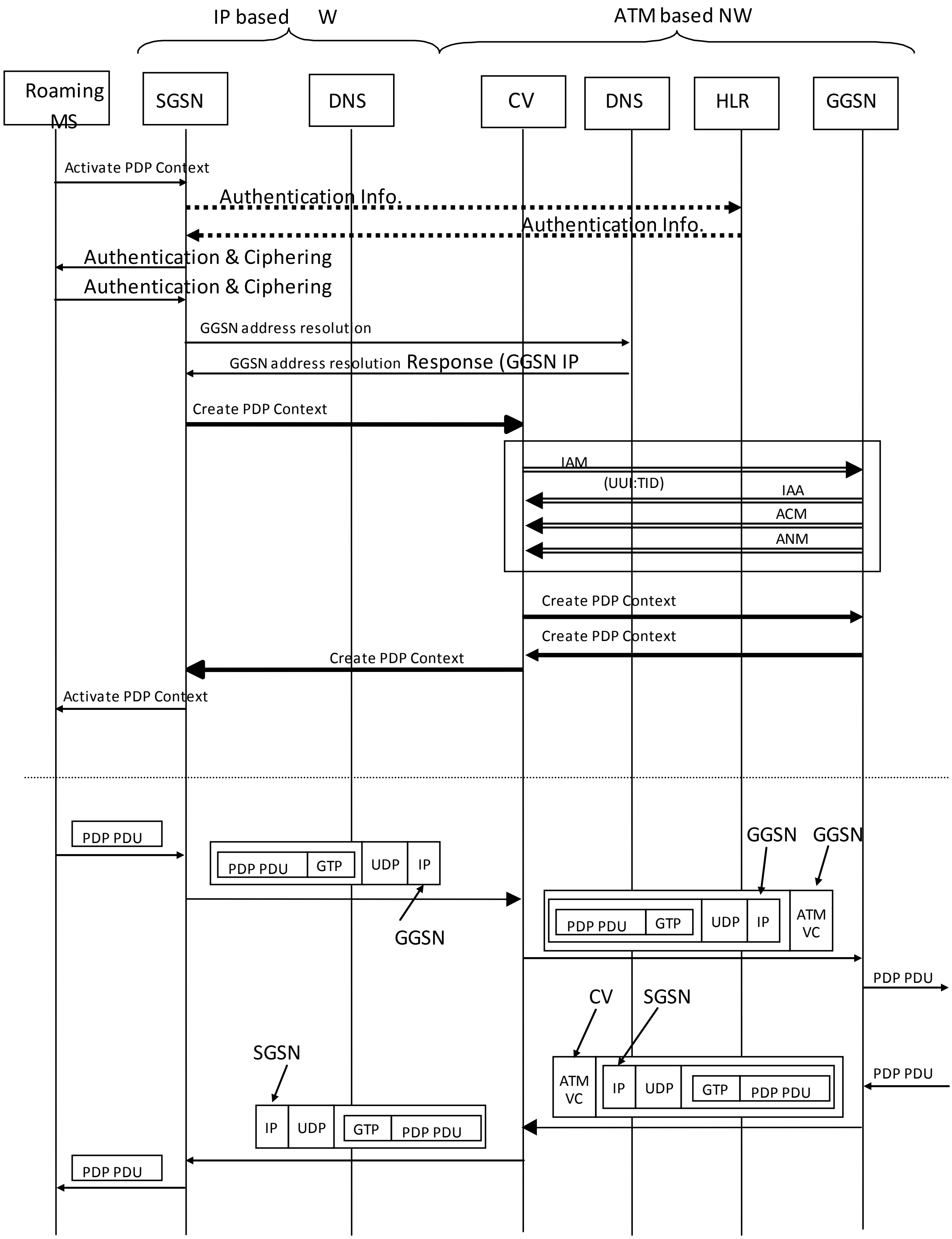
(D) Handover between SGSNs (Different CV)

Each case is shown twice for IP base network originating and ATM SVC base network origination pattern.

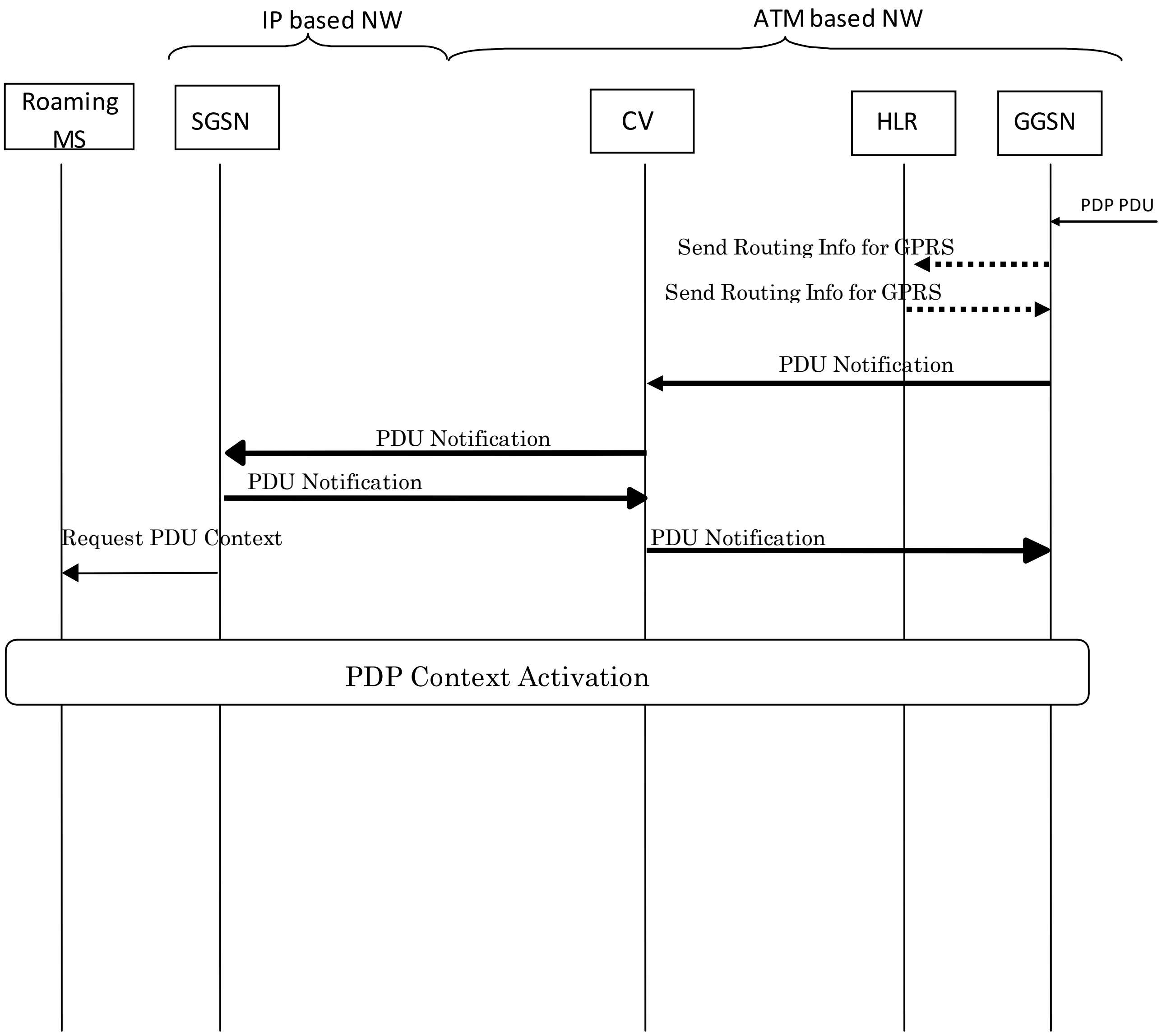
In the figures, CV means the transports layer converter, which maps the node IP address to the ATM address. CV also means the terminating point for the ATM SVC link. The owner of the CV may be the operator of ATM SVC tunneling network or the operator of IP tunneling network (or other). CV manages the SVC link for each GTP tunneling, relaying user packets between IP base tunneling network and ATM SVC tunneling network.

All the messages relating to a particular GSN from one IP base-tunneling network necessary pass the same (logical) CV. This is achieved by setting the IP routing tables (by means of IP routing protocol sent out from the CV) such that all the messages above pass the same CV. Note that one UMTS network may have multi (logical) CV for the particular GSN. This is because that the UMTS network may have some connecting networks, which are not logically connected by any way.

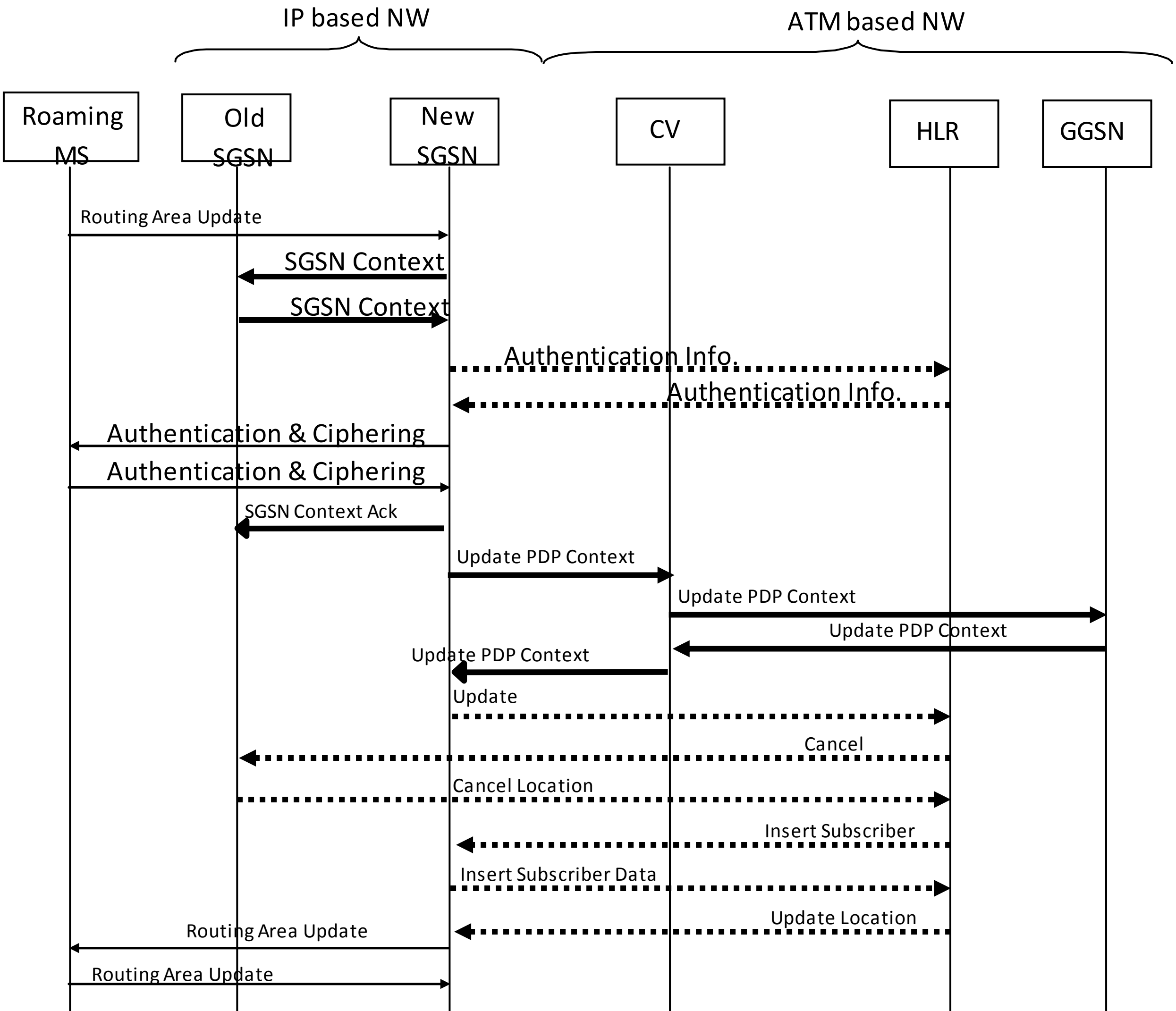
(1)Originating Call



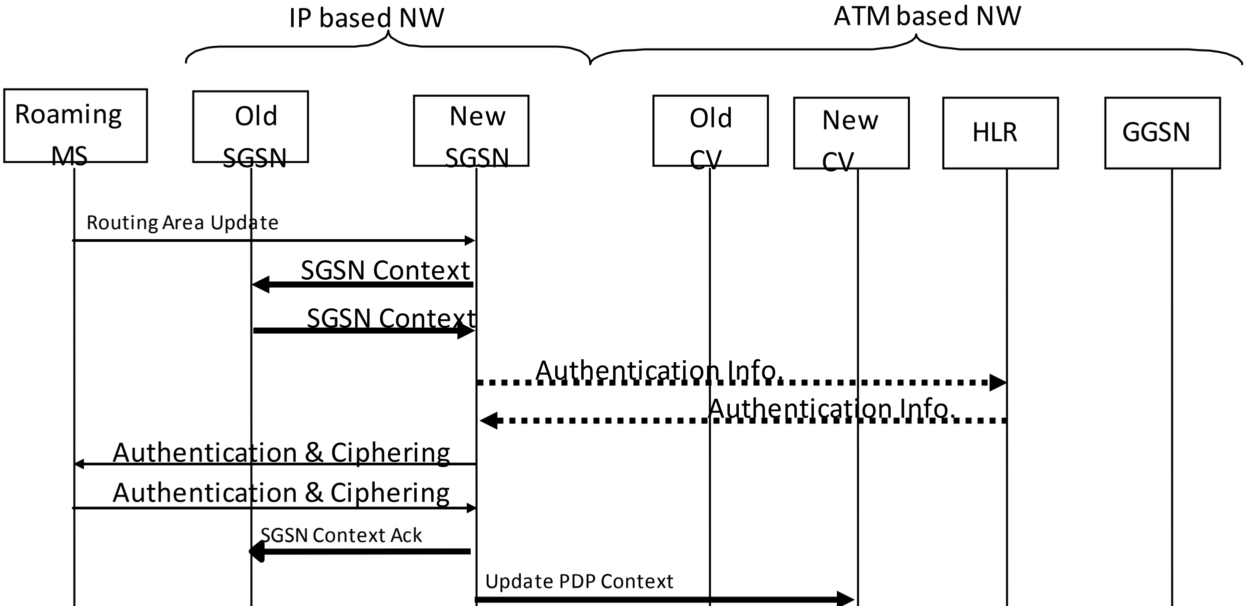
(2)Terminating Call



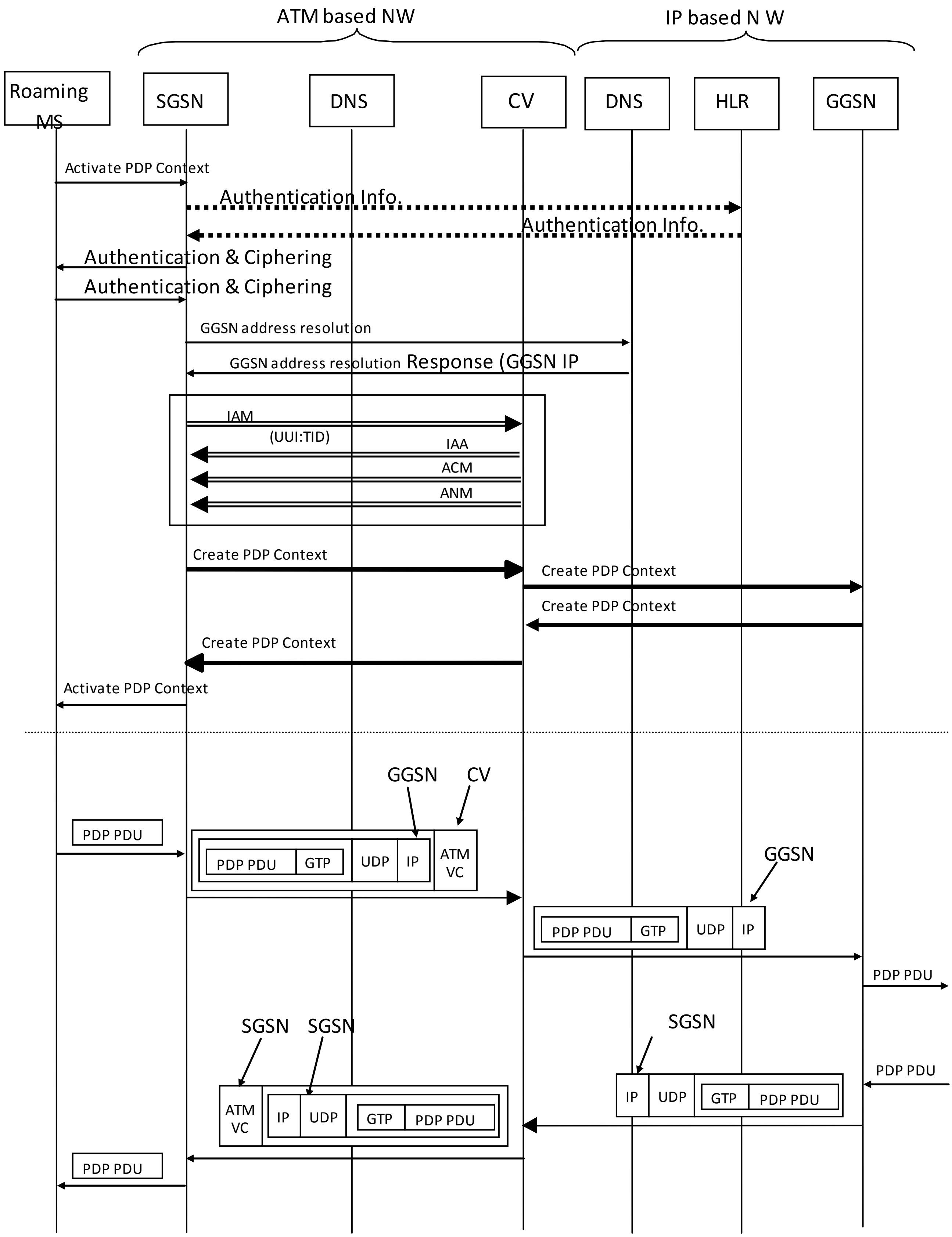
(3)Handover between SGSNs (Same CV)



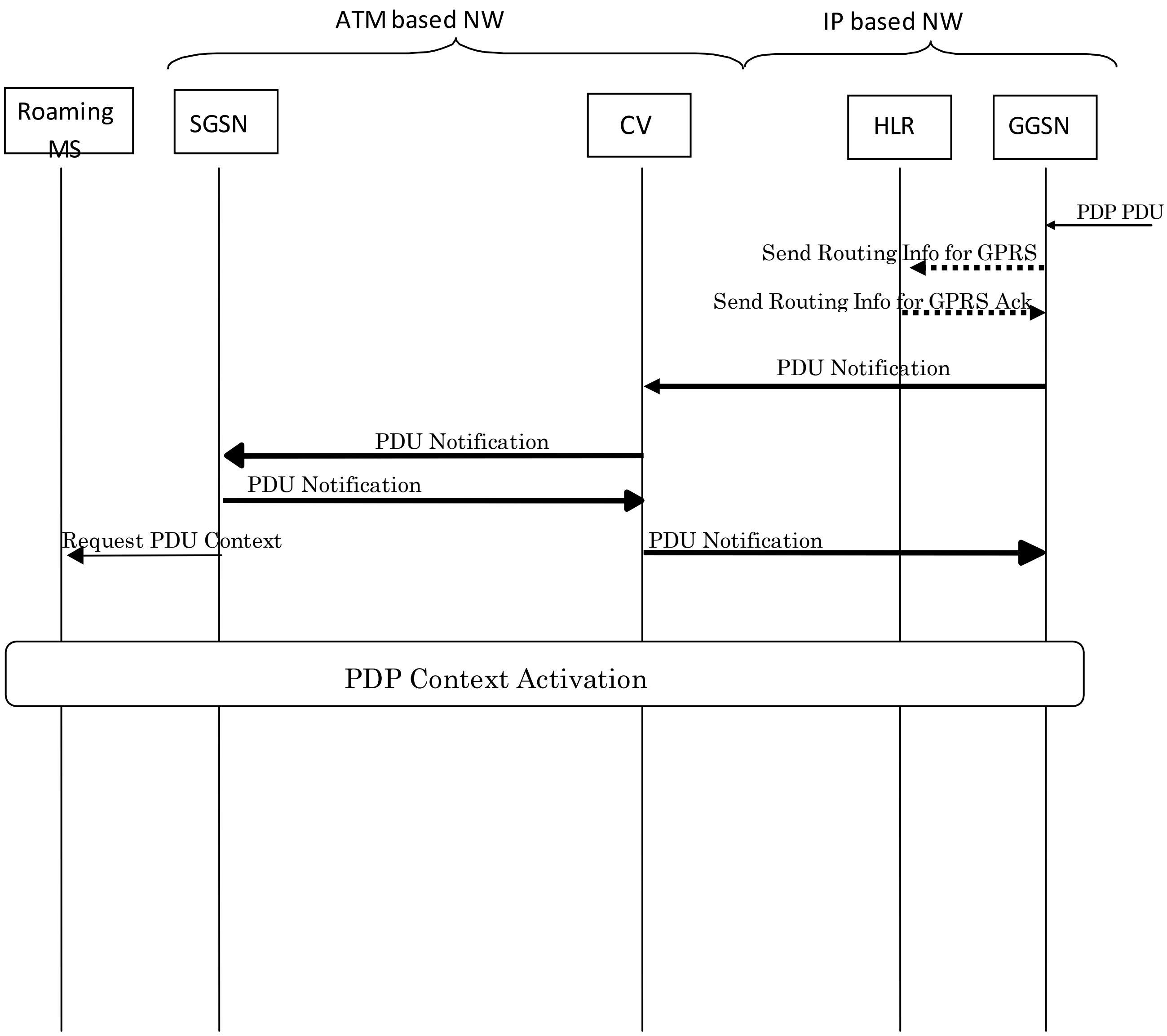
(4)Handover between SGSNs (Different CV)



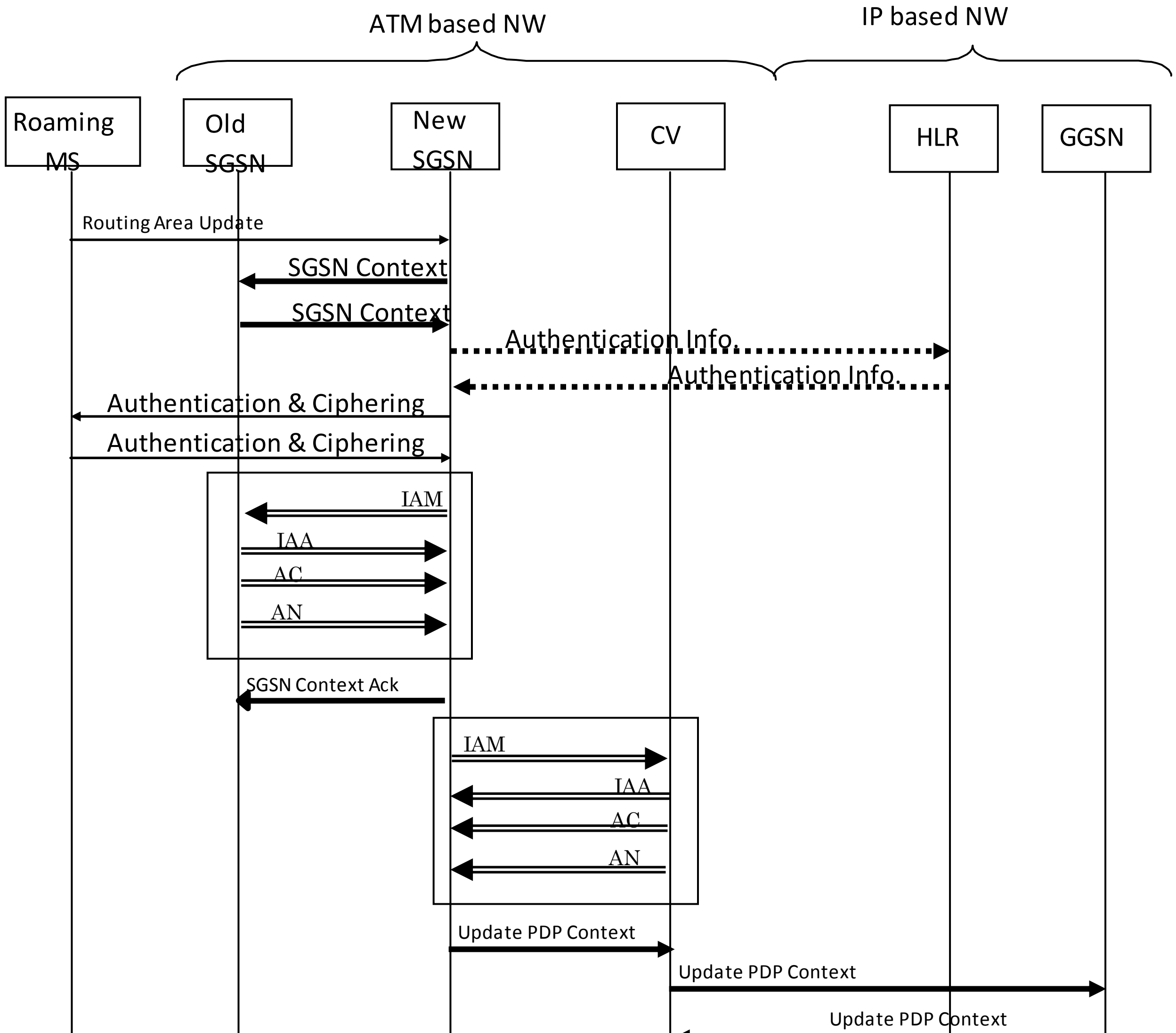
(5)Originating Call



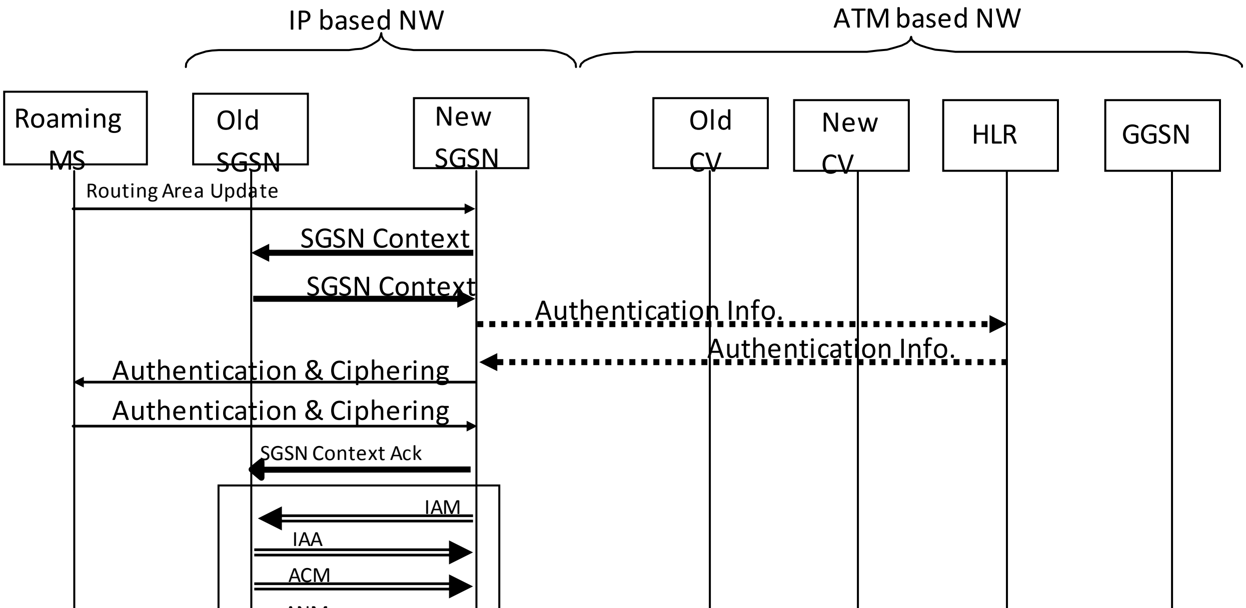
(6)Terminating Call



(7)Handover between SGSNs (Same CV)



(8)Handover between SGSNs (Different CV)



History

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