IMS   All-IP Network   Conversion of CS to IP-based Network

●Technology Reports●

Special Articles on All-IP Network Technology —Evolution of Core Network —

IP-based FOMA Voice Network toward Enhanced Services and Improved Efficiencies

NTT DOCOMO is making efforts to convert its CS core network to an IP-based network in order to realize an All-IP network. Currently, the FOMA network is providing services to more than 52 million users. Therefore, the basic requirements set for the development of the IP-based core network are a) to continue providing the present voice services and, b) not to have any negative effect on the existing 3G radio access network and 3G terminals. IMS nodes and their gateway node to the 3G radio access network have been developed with the objectives of satisfying the above requirements and, in addition, accommodating LTE and establishing a service infrastructure that enables the adding of value in the future.

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1. Introduction

Mobile communication services and their contents are becoming richer and richer these years as a result of the sophistication of the terminals and the spreading of flat-rate tariff plans. As a consequence, the mainstream of communications traffic is shifting from Circuit Switched (CS) type to Packet Switched (PS) type and it is forecast that PS type traffic will continue to grow as a result of further service diversification. NTT DOCOMO has been introducing IP technologies in a stepwise manner responding to the changing traffic characteristics, and has already converted the PS domain providing the PS functions to an IP-based network [1]. The next objective is to convert the Asynchronous Transfer Mode (ATM)[*1]-based CS domain that currently provides CS services such as voice to an IP-based network. The advantage of converting the network lies in the fact that the overlap in equipment investment in the ATM-based CS domain and the IP-based PS domain can be avoided and that a network can be constructed which can economically cope with communications traffic increases. In addition, by providing both CS and PS services on the same IP network it will be possible to provide services that interact between voice and the Web in an effective and responsive

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*1 **ATM**: A communication scheme in which fixed-length frames called cells are transferred successively.

manner. This will complete the implementation of an All-IP core network (**Figure 1**). Furthermore, as LTE[*2] is introduced in the future, further increases in packet communication speeds can be achieved.

However, because voice is a real time service, higher quality performance has been required for the CS domain than that for the PS domain. Therefore, it is a technical challenge to achieve the same voice service quality over IP as that which has been provided over ATM networks.

NTT DOCOMO has chosen to migrate the CS domain from an ATM-based core network to an IP-based core network (hereinafter referred to as "CS-IP NW") that controls and transmits voice traffic using IP Multimedia Subsystem (IMS)[*3] as its approach to introducing IP technologies to the voice services that have been provided over CS [2]. With IMS standards, the assumption is that the terminals are equipped with the Session Initiation Protocol (SIP)[*4] communication function, but the 3G terminals in use in NTT DOCOMO's

3G-CS network, which already provides voice services to 52 million users (as of January 2010), do not have this SIP communication function. In the course of the migration, it is important that existing 3G terminals and 3G radio access network interface will not be affected from the viewpoint of the impact on users.

This article describes the methods developed to solve these technical issues and also presents an overview of NTT DOCOMO's CS-IP NW architecture and basic sequence flows as well as
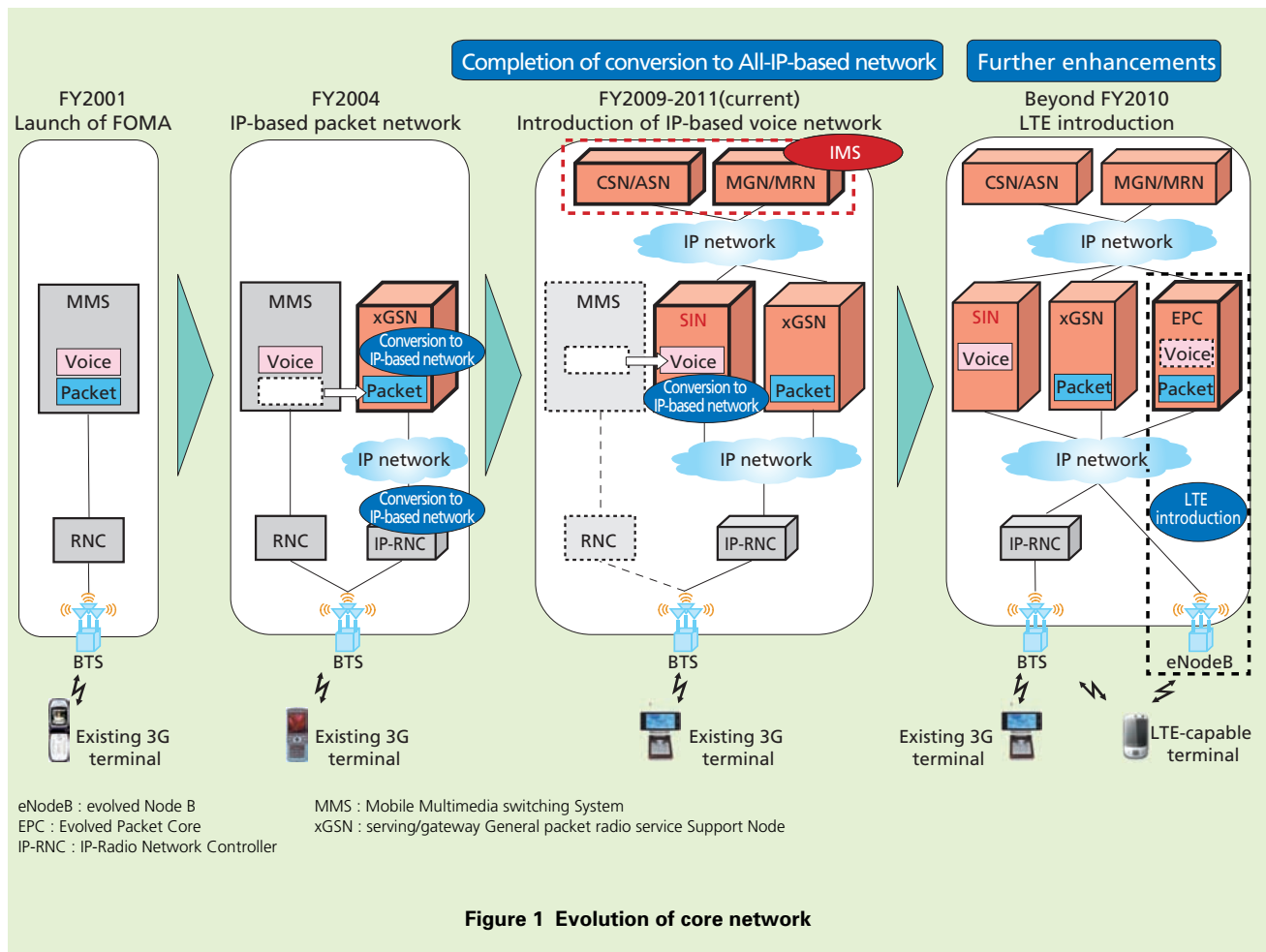

eNodeB : evolved Node B
EPC : Evolved Packet Core
IP-RNC : IP-Radio Network Controller
MMS : Mobile Multimedia switching System
xGSN : serving/gateway General packet radio service Support Node

Figure 1 Evolution of core network

*2 **LTE**: Extended standard for the 3G mobile communication system studied by 3GPP. It is equivalent to "3.9G" or Super3G as proposed by NTT DOCOMO.

*3 **IMS**: A communication system standardized by 3GPP for achieving multimedia services by integrating communication services of the fixed-line network, mobile communications network, etc. using IP technology and SIP (see *4) protocol as used in VoIP.

*4 **SIP**: A call control protocol defined by the Internet Engineering Task Force (IETF) and used for IP telephony with VoIP, etc.

future perspectives.

2. Overview of CS-IP NW Architecture

Figure 2 (a) shows the architecture of the CS-IP NW that has been developed.
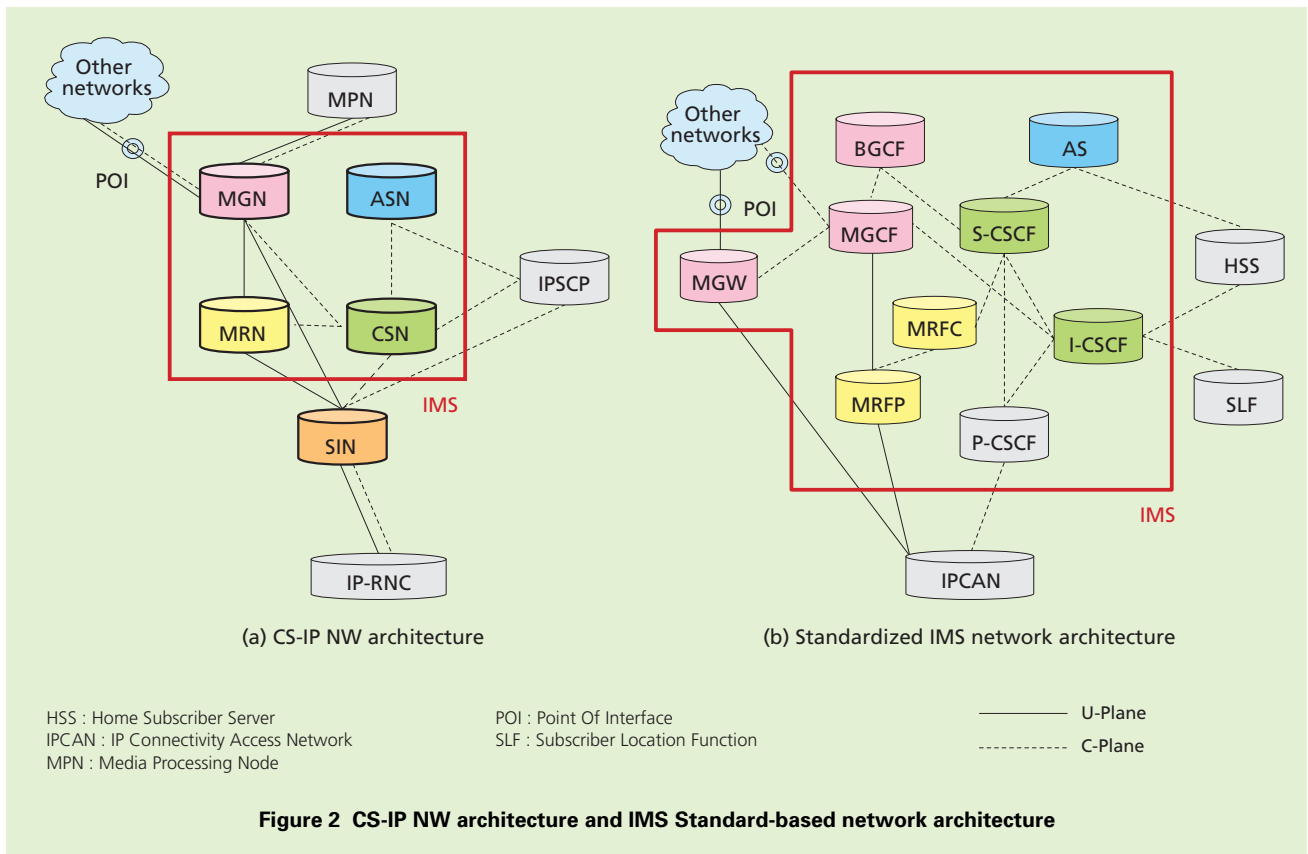
The Call Session control Node (CSN), Application Serving Node (ASN), Media Gateway Node (MGN) and Media Resource Node (MRN) perform session control, service control, other network connection controls, and announcement control functions respectively. Their functions are the same as those in the standardized IMS. In the standardized architecture of IMS shown

in Fig. 2 (b), the CSN corresponds to the Interrogating/Serving-Call/Session Control Function (I/S-CSCF), the ASN to the Application Server (AS), the MGN to the Breakout Gateway Control Function (BGCF)/Media Gateway Control Function (MGCF)/the Media Gateway (MGW), and the MRN to the Media Resource Function Controller (MRFC)/Media Resource Function Processor (MRFP).

In migrating the 3G-CS network to the CS-IP NW, any possible impact on existing networks has to be solved. In the CS-IP NW, a Signaling Interworking Node for 3G access (SIN) has been developed as the gateway equipment to

accommodate the 3G radio access network in the IMS network. The CS-IP NW has to provide the 3G radio access network with the same functions as the existing 3G-CS network does, in which case the SIN has the role of connecting the existing 3G radio access network to the IMS [3]. The SIN acts as a virtual terminal having a SIP function in place of 3G terminals that do not have the SIP communication capabilities, and connects to the CSN that is the equivalent of the I/S-CSCF.

The ASN works as an AS that provides various network services such as answering phones and call forwarding which are provided by the existing

(a) CS-IP NW architecture   (b) Standardized IMS network architecture

HSS : Home Subscriber Server
IPCAN : IP Connectivity Access Network
MPN : Media Processing Node

POI : Point Of Interface
SLF : Subscriber Location Function

——— U-Plane
----------- C-Plane

Figure 2 CS-IP NW architecture and IMS Standard-based network architecture

3G-CS network, so that the users in the CS-IP NW are provided with the same services in the same manner.

3. Functional Characteristics of CS-IP NW

3.1 Inheritance of Interfaces with 3G Radio Access Network

An ordinary IMS-capable terminal and an existing 3G terminal have been compared from protocol point of view (**Figure 3**). Here, the actual conversion processing by the SIN in the case of originating/terminating call control is shown.

In the originating call control of an IMS-capable terminal, direct communication using SIP protocol with the IMS equipment is possible because the terminal has the SIP protocol communication function. Whereas in the case of an originating call from an existing 3G terminal, the call is initiated by the Call Control (CC) protocol used between the 3G-CS network and the 3G terminal. At the SIN, when a CC protocol signal is received, it converts the signal to a SIP signal and sends it to the IMS. In the case of call termination control, SIP is converted to CC in the opposite way. In such a manner, even if the existing 3G terminal does not support SIP, call control at the IMS is made possible by the protocol conversion processing carried out at the SIN.
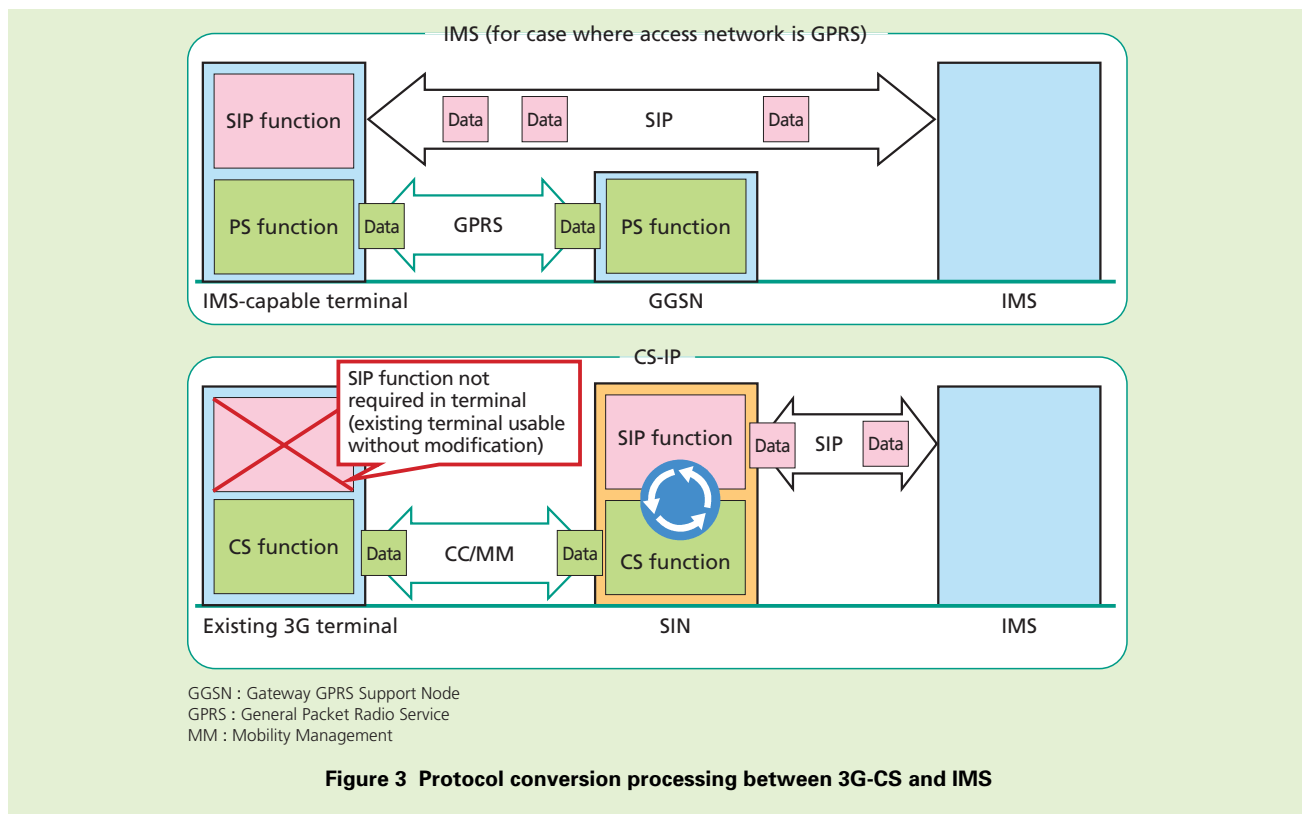
In addition to the above protocol conversion, the SIN performs such functions as authentication, ciphering and radio access bearer assignment necessary for the radio access network.

3.2 Ensuring Voice Quality in CS-IP NW

1) Bandwidth Management at Call Connection Time

In order to cope with the explosive increase in PS type traffic, NTT DOCOMO has been expanding its IP backbone transmission network. This time, with the conversion of the 3G-CS network to an IP-based network, the CS type traffic will also be carried

GGSN : Gateway GPRS Support Node
GPRS : General Packet Radio Service
MM : Mobility Management

Figure 3 Protocol conversion processing between 3G-CS and IMS

by the IP backbone in addition to the existing PS type traffic. Telecommunication operators usually calculate and secure necessary bandwidth resources based on demand forecasts, but in the core network of a mobile network, communications traffic depends heavily on the behavior of the users. For example, the traffic of originating and terminating calls fluctuates considerably when large groups of people gather at events. Because of this, the communications traffic generated can exceed the bandwidth resources that had been prepared, and this may result in communication quality deterioration such as delay and data loss.

In order to realize QoS[*5] guaranteed type services, of which voice is a typical example, on an IP network, it is advantageous to make use of the CS concept according to which a connection is established after the necessary end-to-end bandwidth is secured. This requires an architecture that will inhibit the inflow of communications traffic exceeding the bandwidth resource that has been secured.

There exists the Resource reservation Protocol (RSVP)[*6] as a bandwidth management function for an IP network. In this protocol, because each router maintains state information per flow, when the size of the network becomes big, the number of flow states that has to be managed increases. Furthermore, another drawback is that the delay is increased because the flow state has to be referred to before forwarding the packets, which makes it inappropriate to be applied to a large network.

Based on these considerations we have adopted a bandwidth management scheme for the CS-IP NW whereby the bandwidth management is performed by the CSN for each section. Its operation is shown in **Figure 4**.
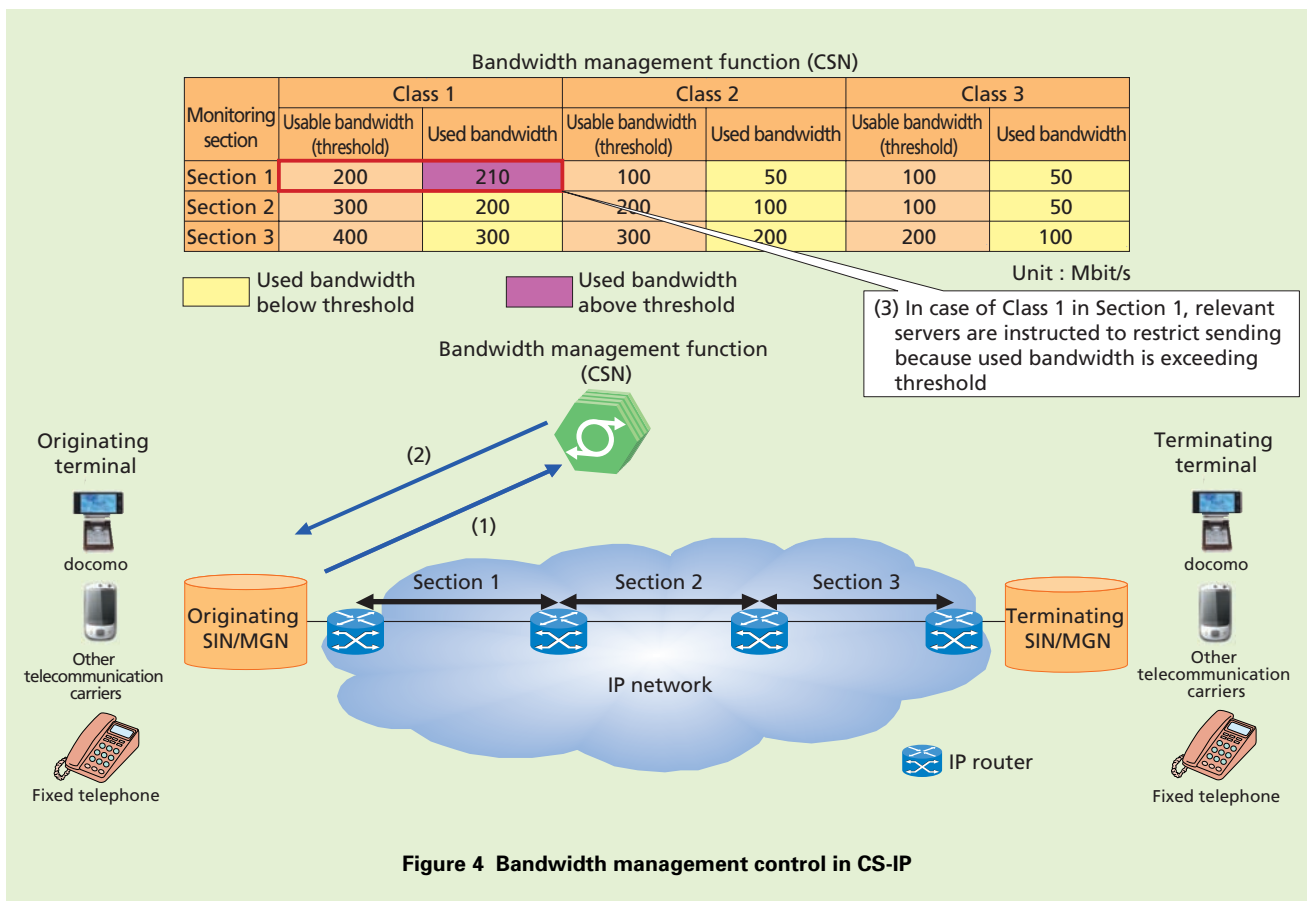

Bandwidth management function (CSN)

Monitoring section	Class 1		Class 2		Class 3	
	Usable bandwidth (threshold)	Used bandwidth	Usable bandwidth (threshold)	Used bandwidth	Usable bandwidth (threshold)	Used bandwidth
Section 1	200	210	100	50	100	50
Section 2	300	200	200	100	100	50
Section 3	400	300	300	200	200	100

Unit : Mbit/s

☐ Used bandwidth below threshold   ☐ Used bandwidth above threshold

(3) In case of Class 1 in Section 1, relevant servers are instructed to restrict sending because used bandwidth is exceeding threshold

Figure 4 Bandwidth management control in CS-IP

(1) The SIN/MGN that first receives an originating call request from a calling user (hereinafter referred to as "Originating SIN/MGN") sends the destination/source IP addresses, data bandwidth and priority information (hereinafter referred to as "Priority Class") on a per-call basis to the CSN having bandwidth management function. The Priority Class is obtained by classifying the priority of the call on the basis of whether, for example, it is an emergency call or a normal call.

(2) The CSN calculates the bandwidth to be used in each section by sorting and summing up all the information received in (1) on a per-Originating SIN/MGN and Terminating SIN/MGN basis. Here, the "Terminating SIN/MGN" refers to the SIN/MGN that makes terminating requests to the terminating user.

(3) The CSN having the bandwidth management function sets the available bandwidth (threshold) for each Priority Class and section, and if the bandwidth calculated in (2) above exceeds the threshold, it restricts new calls by sending back an error response to the Originating SIN/MGN.

With these bandwidth management functions, it is possible to restrict incoming flows of traffic greater than the available bandwidth and to ensure the QoS of the voice calls in progress.

2) QoS Monitoring Function after Connection Establishment

A QoS monitoring method has been introduced in the CS-IP NW in order to cope with quality degradations caused by failures of multiple routers and/or silent failures[*7] after a connection is established. Specifically, this method detects and takes actions against quality degradations which are caused by router network congestion and/or router failures in the User Plane (U-Plane)[*8] route used for communication, by exchanging Real-time Transport Control Protocol (RTCP)[*9] packets for QoS monitoring among SIN, MGN and MRN and to monitor loss and delay of these RTCP packets. The detailed operationof the QoS monitoring is described below.

The originating node separates the user data traffic (hereinafter referred to as "user traffic") and the RTCP packets for QoS monitoring purposes, and then assigns the user traffic an identifier with a higher priority than that for RTCP packets.

Figure 5 (a) shows the case where the traffic load on the IP backbone network is low. In this case, because there are still enough bandwidth resources for the user traffic, packets arrive from the originating node to the terminating node without any loss. However, when congestion in the router network of the
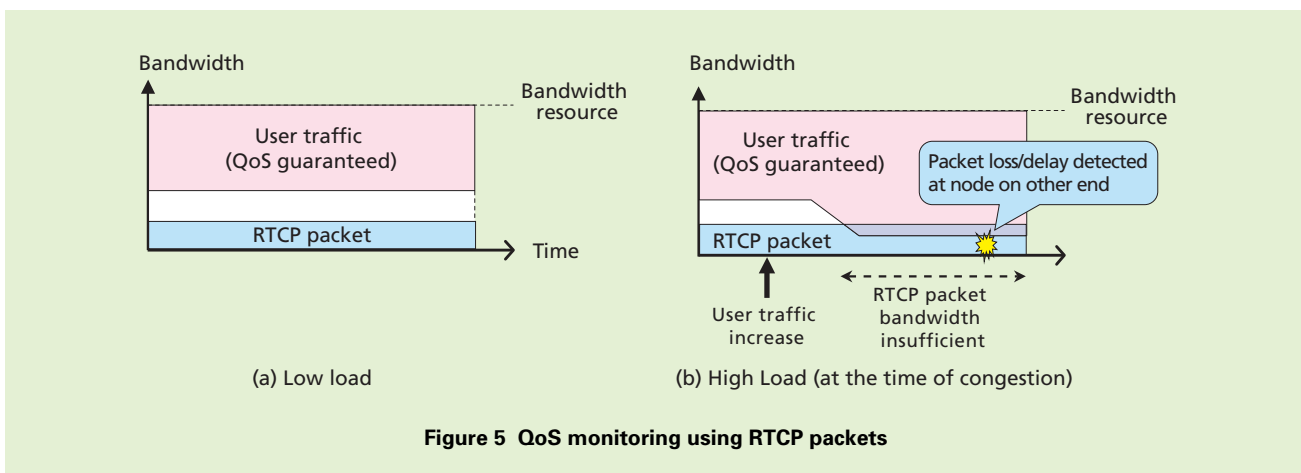

Figure 5 QoS monitoring using RTCP packets

IP backbone occurs and the sum of the user traffic and the RTCP packets exceeds bandwidth resources, the RTCP packets having lower priority start to experience loss/delay as a result of the priority control in the routers in the IP backbone (Fig.5 (b)). RTCP packet losses also occur when the routers in the IP backbone network stop communication due to silent failures or other reasons. In such cases, when the terminating node cannot receive RTCP packets from the opposite node within a certain fixed monitoring time, it decides that there is a QoS degradation or that the user message is not conveyed during the connection

The number of QoS degraded calls and calls where packets are lost within a unit of time is also monitored and if the number exceeds a certain level it will generate an autonomous message to alert the maintenance personnel.

In this way, by interpreting the non-arrival of RTCP packets, including loss and delay within the monitoring time, as a precursor of user traffic QoS degradation, it becomes possible to swiftly initiate failure analysis.

4. Basic CS-IP Control Method

4.1 Location Registration Control

When location registration is requested from a 3G radio access network, the SIN chooses a CSN that has already been registered by asking the IP Service Control Point (IPSCP) and sends the SIP_REGISTER message to this CSN (note that the first location registration is performed in a round-robin[*10] fashion). After this the IMS registration is performed among the SIN, CSN, ASN and IPSCP and the profile is kept within each node.

4.2 Basic Originating/terminating Call Control

Figure 6 shows the basic originating/terminating call control procedure that follows 3G location registration.

1) Control on Originating Side

The salient points about the call control on the originating side are the interworking between the 3G radio access network and the IMS at the SIN, and the bandwidth management performed at the CSN in order to guarantee the QoS of voice calls.

Following call initiation by the originating terminal, the terminal and the SIN start the process of authentication and ciphering (Fig.6 (1)). The originating SIN that has received call initiation request from the terminal sends back an acknowledgement and at the same time selects from the subscriber profile the address of the CSN that accommodates the originating user and
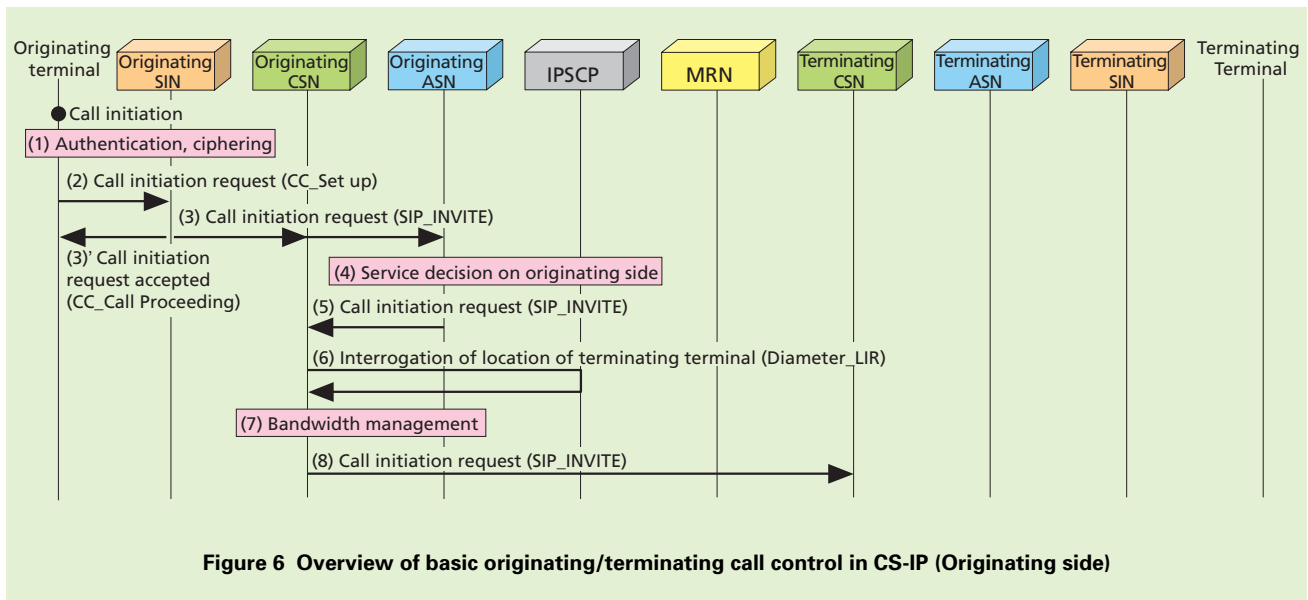

Figure 6 Overview of basic originating/terminating call control in CS-IP (Originating side)

sends a call initiation request to this CSN (originating CSN). On receiving this request the CSN identifies the ASN from the initial Filter Criteria (iFC)[*11] and starts performing the originating call request (Fig.6 (2)(3)(3)'). The originating ASN identifies the originating user utilizing the user ID contained in the call initiation request and after checking the service conditions on the originating side by referring to the user profile, makes an originating call request to the originating CSN (Fig.6 (4)(5)).

The originating CSN accesses the IPSCP that accommodates the terminating user and retrieves the address of the CSN where the terminating user is registered (Fig.6 (6)). The originating CSN calculates the bandwidth to be used based on the destination/source IP addresses, data bandwidth and Priority

Class and if the bandwidth is within the allowable bandwidth (threshold) makes a call initiation request to the terminating CSN (Fig.6 (7)(8)).

2) Control on Terminating Side

The terminating CSN identifies the terminating user utilizing the terminating Uniform Resource Identifier (URI) contained in the call request from the originating CSN and makes a call initiation request to the terminating ASN (**Figure 7** (1)). The terminating ASN, upon its receipt, checks the service conditions on the terminating side including the contractual status of the called user, and then sends a call initiation request to the terminating CSN (Fig.7 (2)(3)). The terminating CSN calculates the bandwidth to be used based on the destination/source addresses, data bandwidth and Priority Class and if the bandwidth is within the allowable band-

width (threshold) makes a call initiation request to the terminating SIN where the terminating user is registered (Fig.7 (4)(5)). The terminating SIN sends a call initiation request to the terminating terminal after performing paging processing, authentication and ciphering (Fig.7 (6)(7)) and at the same time sends a provisional response to the originating SIN assigning capability information concerning the session (Session Description Protocol (SDP)) such as the address of the terminating SIN and the codec (Fig.7 (8)). The terminating terminal that has received the call initiation request from the terminating SIN sends back its acknowledgement to the call initiation request (Fig.7 (9)).

3) Tone Sending Control

The tone sending control has the function of controlling transmission of the Ring Back Tone (RBT) to the user.

Figure 7 Overview of basic originating/terminating call control in CS-IP (Terminating side)
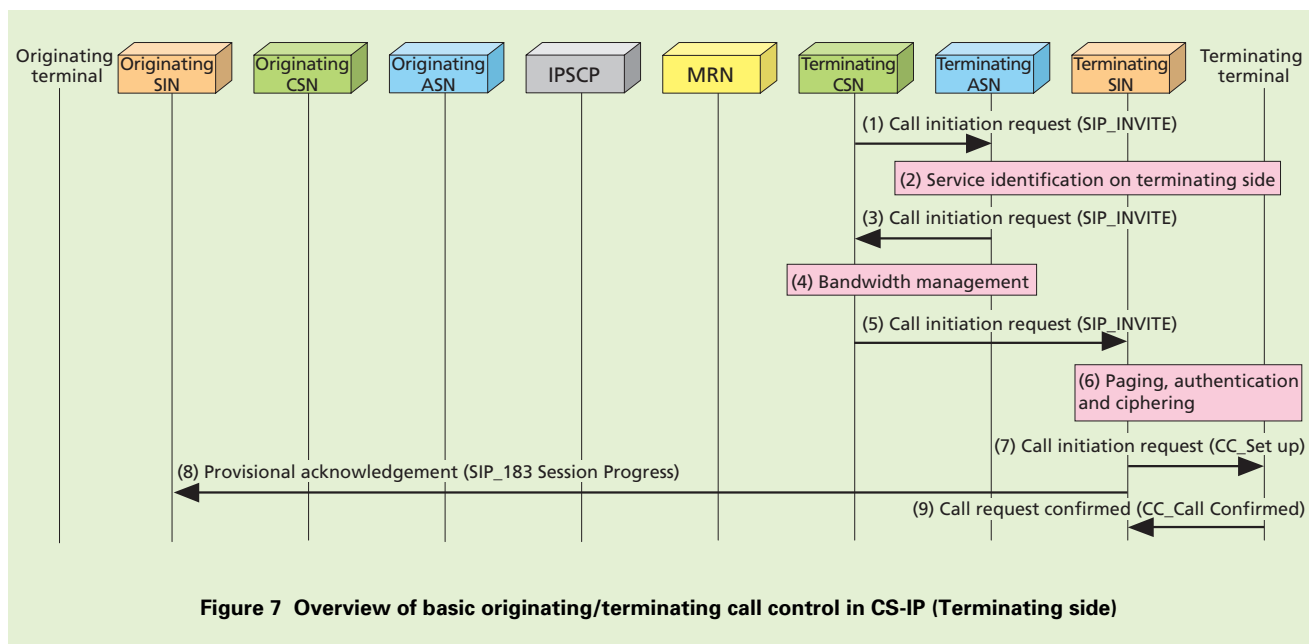
*11 **iFC**: Information containing a list that assists deciding which AS to select in order to provide services to the user.

One of its features is that the terminating ASN selects the MRN closest to the originating side in order to optimize the U-Plane route.

The terminating SIN that received the ringing signal from the terminating terminal sends the ringing signal to the terminating ASN via the terminating CSN (**Figure 8** (1)(2)). The terminating ASN, after having received the ringing signal from the terminating CSN, checks the RBT connection and makes a connection request via the terminating CSN to the MRN which is the source of the tone (Fig.8 (3)(4)). In response, the MRN sends to the terminating ASN via the terminating CSN a connection request response that assigns the SDP for sending the RBT (Fig.8 (5)). The terminating ASN confirms the RBT connection and sends to

the originating SIN an updating signal assigning the SDP of the MRN, so that the SDP exchange is changed from between the originating and terminating SINs to between the originating SIN and MRN (Fig.8 (6)). After the SDP updating is completed, the terminating ASN sends the ringing signal to the originating ASN, and the originating ASN sends the ringing signal to the originating SIN via the originating CSN (Fig.8 (7)(8)). In addition, the originating SIN sends the ringing signal to the originating terminal (Fig.8 (9)). In parallel to the above, the terminating ASN sends back an acknowledgment to the connection request response from MRN and the MRN that has received this sends the RBT to the originating side (Fig.8 (7)'(8)').

4) Response Control on Terminating Side

The terminating SIN receives the connection acknowledgement from the terminating terminal when the terminating user answers the call (off hook). Then, the terminating SIN replies back with a connection acknowledgement to the terminating terminal and at the same time sends the connection acknowledgement to the terminating ASN via the terminating CSN (**Figure 9** (1)(2)(2)'). Upon receiving acknowledgement from the terminating CSN, the terminating ASN recognizes release of the RBT and requests RBT release via the terminating CSN to the MRN, i.e. the audio source (Fig.9 (3)(4)). The MRN which has received the RBT release request from the terminating CSN responds to the terminating ASN
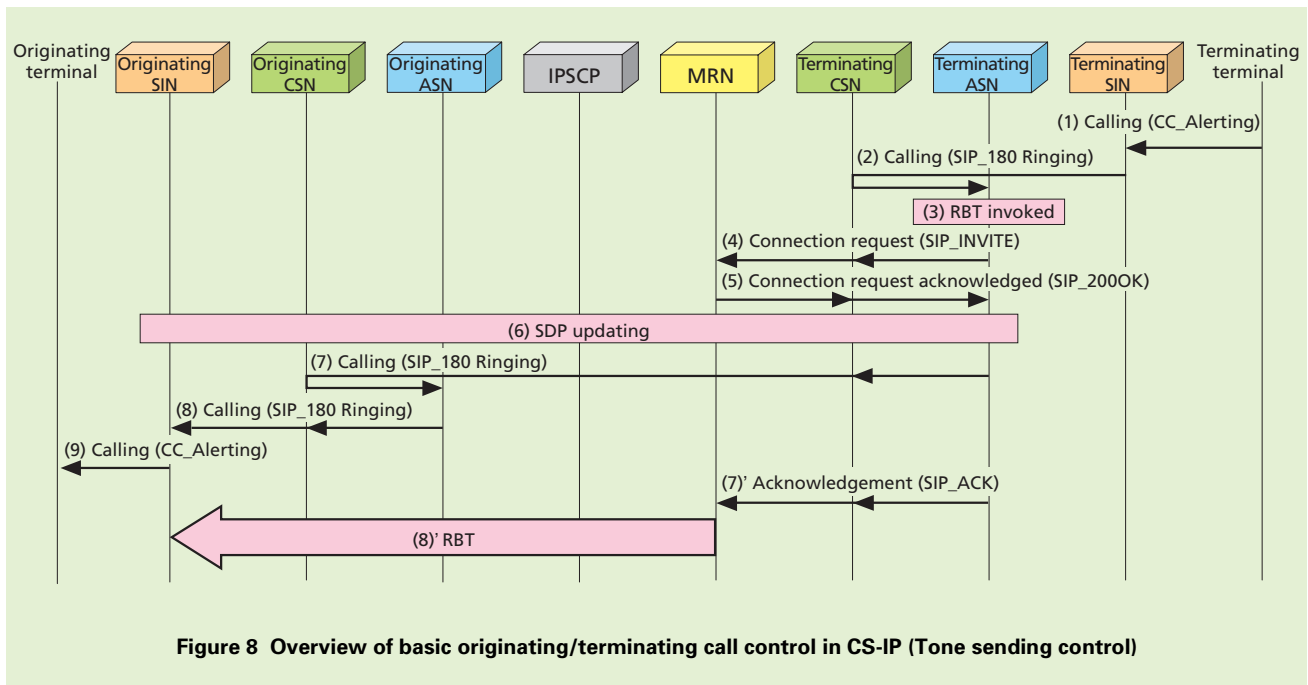

Figure 8 Overview of basic originating/terminating call control in CS-IP (Tone sending control)
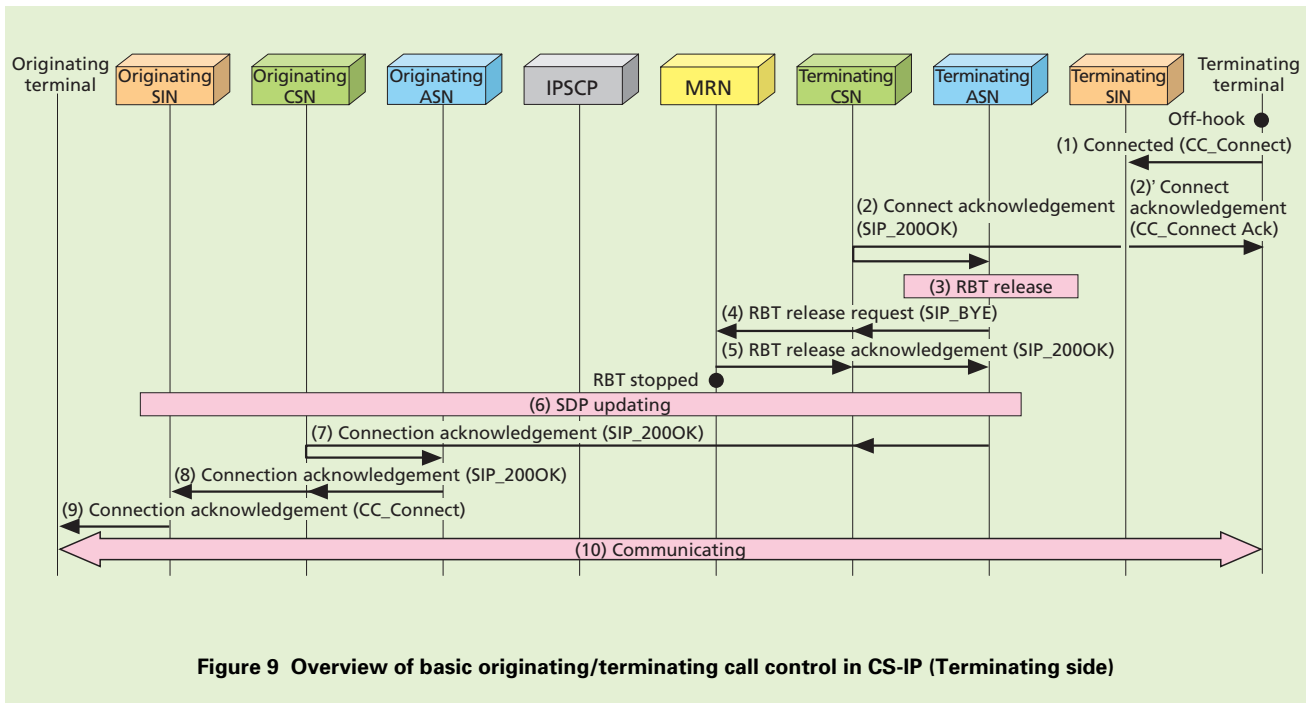
Figure 9 Overview of basic originating/terminating call control in CS-IP (Terminating side)

via the terminating CSN and stops sending the RBT to the originating terminal (Fig.9 (5)). Following that, the originating ASN sends to the originating SIN an updating signal assigning the SDP of the terminating SIN, so that the SDP exchange is changed from between the originating SIN and the MRN to between the originating and terminating SINs (Fig.9 (6)).

After the SDP updating is completed, the terminating ASN sends the connection acknowledgement to the originating ASN via the terminating CSN, and the originating ASN that has received this response connection responds to the originating SIN via the originating CSN. Furthermore, the originating SIN sends the connection acknowledgement to the originating ter-

minal (Fig.9 (7) to (9)).

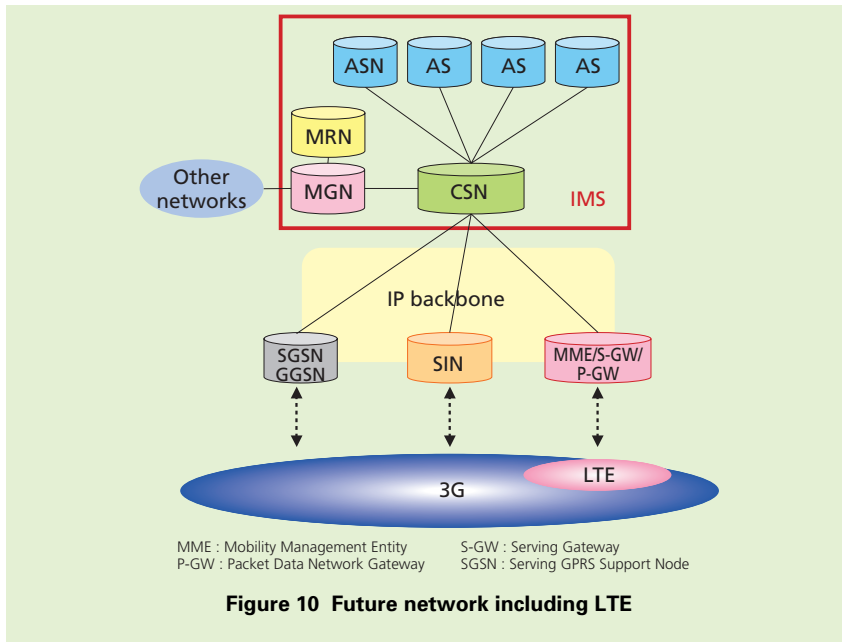Following these steps, it is now possible to communicate over CS-IP.

5. Support of Future Services

In its current form the CS-IP NW is equipped only with an ASN that functions as an AS that provides existing voice services. Studies are currently underway to implement more than one AS, each providing a different service, so that services such as the presence service in the Rich Communication Suite (RCS)[*12] and instant messaging services that involve sophisticated processing can be realized in the future. This will enable the enhancement of flexibility in the provision of services by combining multiple ASs, each of

which is dedicated to a particular service, and thereby facilitating the creation of new services.

Moreover, from the viewpoint of providing services more efficiently, in addition to the accommodation of the 3G-CS network by the SIN presented in this article, interconnection with the 3G packet network and LTE is also being studied in the scope of the IMS (**Figure 10**). In the case of 3G packet network, conversion to the IP-based network is completed, but, by accommodating it into the IMS, it will be become possible to provide IMS services such as RCS common to LTE and 3G packet network that will enable users to receive seamless services without noticing the differences in the access networks.

MME : Mobility Management Entity
P-GW : Packet Data Network Gateway
S-GW : Serving Gateway
SGSN : Serving GPRS Support Node

Figure 10 Future network including LTE

6. Conclusion

This article has described the development work leading to the conversion of the 3G-CS network to an IP-based network. From now onwards we plan to complete the migration of the CS core network to an IP-based network and to expand the services offered by the IMS one by one to other access networks.

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