A QoE based service control scheme for RACF in IP-based FMC networks

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Abstract

Regardless of the fact that convergence is a way to evolve existing fixed and mobile networks towards NGN, the behavior of IP packet transmissions over mobile networks differs considerably from that for fixed networks. In IP-based FMC (Fixed Mobile Convergence) networks, the fluctuation ranges of the network QoS metrics observed by the NGN terminals are remarkably large. In this respect, it is difficult to estimate the real end-to-end QoS metrics and to control the services properly between the NGN terminals, considering the end-user QoE.

In this paper, for FMC services, we provide a service control scheme for RACF (Resource and Admission Control Functions) based on the end-user QoE predicted by the per-segment based observation method of the network QoS metrics using RTP/RTCP [1]. In IP-based FMC networks, the proposed scheme can be applied to all general NGN services provided by RTP/RTCP. The proposed scheme is closely related to the study of ITU-T draft Recommendation Y.MPM [2] and ITU-T Recommendation Y.2111 [3] R2 in ITU-T SG 13.

1. Introduction

The principles of NGN, which are described in ITU-T Recommendation Y.2011 [4], include a variety of service scenarios. For example, a concurrent service that operates a speech session and other sessions such as video, FTP data, IPTV streaming and so on simultaneously, should be more commonly used in NGN. Furthermore, as NGN terminals become more sophisticated, they must be able to provide stress-free operation of multimedia sessions with other different NGN and non-NGN terminals. In short, it can be readily understood that the number of multi-session services that support multiple service sessions should increase remarkably in accordance with the development of NGN.

NGN architecture supports the capability to provide mobility within and between its various access network types and mobility technologies. IP communications are expanding widely into a variety of infrastructures based on fixed networks as well as mobile networks. In such FMC networks, it is a crucial issue to study schemes for maintaining the real end-to-end quality of the service sessions between the NGN terminals considering the performance requirements of each session bundled as a multi-session service. Furthermore, performance requirements might diversify based on the end-user perception of the application-specific quality known as QoE (Quality of Experience) or a conscious decision on the part of the end-user considering the environment in which the NGN terminal is used.

By reviewing the current technologies used for VoIP (Voice over IP) systems and the role of a NGN-RACF implementation and NGN QoS capabilities based on ITU-T draft Recommendation Y.MPM, for example, it is easily recognized that the functions related to the media path controlled by RACF can be applied to realize a quality control scheme for service sessions. Thus it is expected that a new type of service control scheme based on end-user QoE will contribute to the realization of practical service quality that better reflects the perceived quality of the target FMC applications in NGN.

This paper is organized as follows. Section 2 indicates the prediction method of the end-user QoE for the VoIP service and provides an example of the predicted performance of the proposed method using the network QoS metrics. Section 3 presents a notification scheme for the end-user QoE and the QoS metrics in IP-based FMC networks and provides evaluation results of the service control (bit rate control of the speech codec) in section 4. Section 5 gives an example of the service resource control architecture and its configuration. Section 6 provides a typical scenario of FMC services controlled by the proposed service control scheme. Finally, Section 7 provides conclusions.

2. Prediction of end-user QoE for VoIP service

2.1 RTCP based scheme

Most multimedia speech sessions over IP-based networks use RTP/RTCP. The RTCP is an accompanying protocol for RTP to feedback IP-based network conditions from RTP receivers to RTP senders. VoIP systems, for example, are based on a technique wherein a digitized

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speech signal stream is segmented into data blocks and the blocks are transmitted in the form of an IP packet that has an IP/UDP/RTP header followed by a speech signal payload. Then, the RTCP extended reports, called RTCP-XR [5], provide useful monitoring information and a diagnosis of VoIP performance between the NGN terminals. Furthermore, several new block types for IP video metrics have also been studied due to the superiority of RTCP as a firewall friendly protocol.

The most widely used subjective speech quality metric, a kind of QoE metric, is the Mean Opinion Score (MOS) defined by ITU-T Rec. P.800 [6]. In recent years, objective methods have been developed to meet the need to produce a good estimate of the MOS, such as non-intrusive methods that predict the MOS without using a reference speech signal. The method defined by ITU-T Rec. P.564 [7], for example, provides the estimated MOS using a set of the network QoS metrics that can be observed in an IP-layer.

We have developed a method for estimating the end-user QoE that is applicable even in a CPE with poor computational capability, such as cell-phone terminals, without causing the substantial deterioration of the predicted performance as shown in Figure 1.

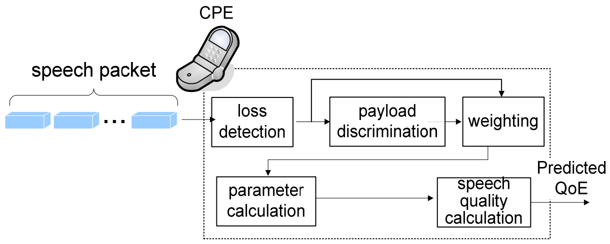

Figure 1. Estimation of QoE in CPE

2.2 Performance analysis of predicted QoE

We conducted performance tests using 880 speech data items. The proposed predicted QoE using RTCP and measured QoE are illustrated in Figure 2. For simplicity, ITU-T Rec.P.862.1MOS-LQO (Mean Opinion Score Listening Quality Objective) [8] is referred to as "measured QoE".

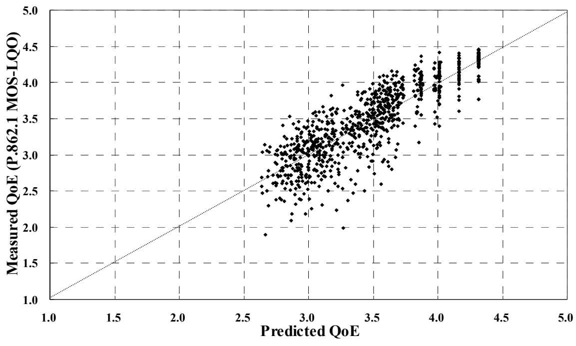

Figure 2. Prediction performance of proposed method

The result shows that the proposed prediction method works well, as the majority of points lie on the dashed diagonal line. The correlation coefficient between the predicted QoE and the measured QoE is higher than 0.86.

3. Notification scheme of the end-user QoE and the network QoS metrics using RTCP

Considering the end-user QoE and the network QoS metrics, to ensure interoperability of the service control scheme and its wide applicability, however, it is of great importance to establish appropriate standards with regards to the required functionality of the equipment. In this section, we present a notification scheme for the QoE and the network QoS metrics using RTCP.

3.1 Observation of RTP streams using RTCP

Several operation scenarios for monitoring session performance indicated by the QoS metrics of networks are discussed. Especially, Mode A, the so-called dynamic operation in ITU-T Rec. P.564, uses information from RTCP-XR packets. The packet provides a set of metrics that contain information for assessing speech quality and diagnosing problems. Figure 3 depicts a configuration of the observation of RTP/RTCP streams.

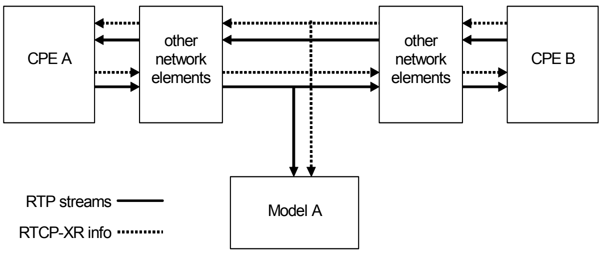

Figure 3. Configuration of RTP streams in ITU-T Rec.P.564

3.2 Identification of RTP and RTCP streams

The monitoring devices should be able to identify RTP/RTCP streams without signaling packets. To identify RTP packets, the following key information is provided. In addition, since one RTCP stream uses a port next to the RTP stream, the RTCP packets are automatically identified by the accompanied RTP packets. The RTP streams generated by each NGN terminal are distinguished by the synchronization source (SSRC) information in the RTP headers.

1) Packet size:
Each codec has a characteristic packet size. Even if a variable bit rate codec is used, the variation in packet size is limited.

2) Port number range:
Although RTP ports are unsettled, they are not perfectly randomized. In many cases, the port number range is limited. They may be fixed in some environments.

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3) RTP header format:
RTP headers include constant and expected parameters such as version(v), padding(p), extension(x), CSRC count(cc), marker(m), and payload type(pt). Multi-RTCP devices can use them to identify RTP packets.
4) Sequence of packets:
RTP packets are sent continuously, incrementing the sequence number with each additional packet. Multi-RTCP devices can exclude scattered packets.

3.3 Per network-segment based notification scheme optimized for IP-based FMC networks

Many mobile networks have retransmission mechanisms to cope with the high bit error rates, but these mechanisms are also associated with deterioration in quality such as large delay variations (jitter) among IP packets. In contrast, the primary cause of jitter on fixed networks is IP packet congestion. Therefore, real-time multimedia applications must deal with the conditions of mobile networks and fixed networks independently if suitable adjustments are to be made. For instance, limiting the sending bit rate improves the total speech quality only in the event of network congestion. For example, a concurrent service which operates a speech session and other sessions such as video, FTP data, IPTV streaming and so on, should be more commonly used in NGN.

The original RTP/RTCP communications use RTCP RR (Receiver Report) to feedback quality conditions of IP-based networks from RTP receivers to RTP senders, i.e., both of the NGN terminals. However, the original RTCP provides overall feedback on the quality of the end-to-end network as a whole, rather than each different network individually. The feedback mechanism provided on an end-to-end network basis is insufficient for controlling the service sessions in FMC networks since the fluctuant quality of end-to-end networks depends on that of both IP-based mobile and fixed networks, and it is difficult to separate the quality of individual networks from the observation of overall quality of end-to-end networks. Hence, we propose a per network–segment based QoE notification scheme using the RTCP known as a Multi-RTCP scheme.

3.3.1 Configuration of Multi-RTCP scheme

The Multi-RTCP scheme supports the scheme for reporting a quality condition of a portion of end-to-end IP-based networks such as FMC networks. The Multi-RTCP scheme identifies the reason for the deterioration in quality using Extra RTCP, which provides the end-user QoE and the QoS metrics of each network divided on an individual basis. The Multi-RTCP scheme is composed of the following components.

1) RTP clients: a kind of NGN terminal
RTP clients support the Multi-RTCP in order to separately report the quality conditions of networks comprising the end-to-end IP-based network in addition to the original RTP/RTCP. They generally have transmission control mechanisms based on the Multi-RTCP Scheme.

2) Multi-RTCP devices based on RACF
Multi-RTCP devices are set at the interconnection points between different networks such as the IP-based fixed and mobile networks. They have network interfaces for monitoring RTP/RTCP streams and the Multi-RTCP communication.

The Multi-RTCP scheme has excellent in compatibility with the original RTCP scheme because RTP senders can be informed of the quality condition of part of the IP-based networks even when all RTP receivers do not support the Extra RTCP Receiver Report. Furthermore the end-user QoE and the network QoS metrics information gathered by Multi-RTCP devices are applicable to the use of network OAM. Figure 4 shows the configuration of the Multi-RTCP scheme.

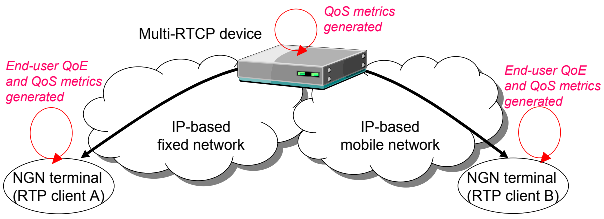

Figure 4. Configuration of Multi-RTCP scheme in FMC networks

3.3.2 Sequence diagram of Multi-RTCP scheme

Each of the RTP clients and Multi-RTCP devices on the routes of the RTP/RTCP exchange Extra RTCP Sender Reports (SR) / Receiver Reports (RR) with a neighboring client/device. The end-user QoE and the QoS metrics of IP-based networks on Extra RTCP RR are concatenated by Multi-RTCP devices and relayed. Figure 5 shows an example of the relay of the Extra RTCP RR. RTP devices receive the Extra RTCP RR, which contains all of the QoE and the QoS metrics created by each of the Multi-RTCP devices.

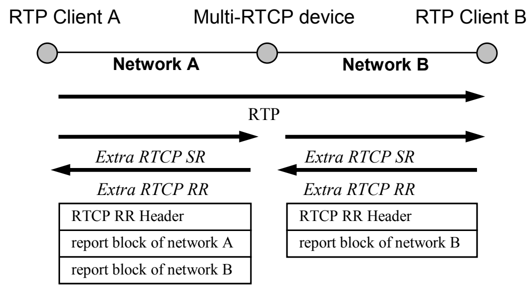

Figure 5. Relay of Extra RTCP RR

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The Multi-RTCP scheme is initiated automatically in accordance with the following two basic rules.

Rule 1: When the Multi-RTCP device detects RTCP SR or Extra RTCP SR, it begins to send Extra RTCP SR to the destination address of the detected packet and Extra RTCP RR to the source address of the detected packet.

Rule 2: When the RTP client or the Multi-RTCP device receives Extra RTCP RR, it begins to send Extra RTCP SR to the source address of the received packet.

Figure 6 shows an example of the initiation of the Multi-RTCP scheme. First, the Multi-RTCP device detects the RTCP SR sent by RTP client A, and then begins to send Extra RTCP SR to RTP client B and Extra RTCP RR to RTP client A following Rule 1. RTP client A, which receives Extra RTCP RR from the Multi-RTCP Device, begins to send Extra RTCP SR to the Multi-RTCP Device following Rule 2. Similarly, RTP client B, which receives Extra RTCP SR from the Multi-RTCP Device, begins to send Extra RTCP RR to the Multi-RTCP Device following Rule 1.

Once the Multi-RTCP scheme is initiated, the original RTCP scheme stops immediately. The number of Multi-RTCP packets is the same as that used for the original RTCP packets. The Multi-RTCP scheme has no impact on the increase in RTCP traffic.

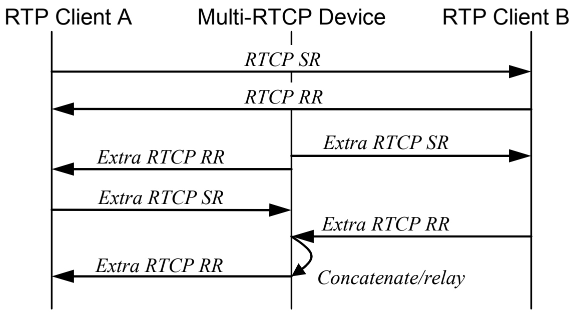

Figure 6. Initiation of Multi-RTCP scheme

3.3.3 Packet format of Multi-RTCP

The format of the Extra RTCP RR is similar to the original RTCP RR consisting of multiple blocks. Extra RTCP RR includes several concatenated report blocks, each of which informs the end-user QoE and the QoS metrics of each part divided by the number of Multi-RTCP devices. An example of the Extra RTCP RR is shown in Figure 7. The majority of the report blocks are the same as for the standard RTCP RR except for the "last SR (LSR)" and "delay since last SR (DLSR)." LSR and DLSR are used for calculating the RTT with the local time of sending Extra RTCP SR and receiving Extra RTCP RR. LSR and DLSR are meaningless without local time and must be replaced with the calculated RTT. When Multi-RTCP devices insert their own QoS metrics into the head of the report block, Multi-RTCP devices replace LSR and DLSR in the relaying report block with the calculated RTT and sender's IP addresses.

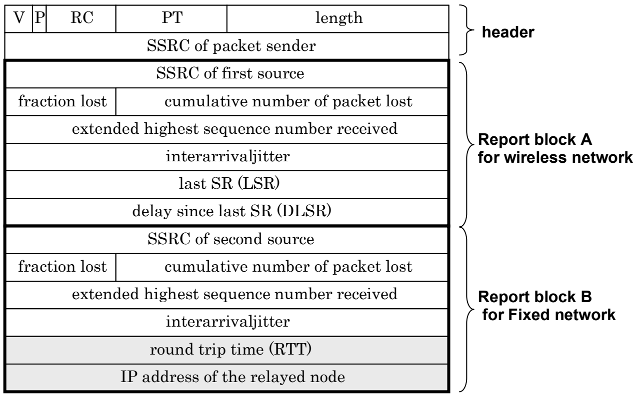

Figure 7. Format of the Extra RTCP RR

4. Example of service control

To evaluate the performance of typical service control such as the bit rate control of the speech codec based on the Multi-RTCP scheme, we developed a prototype system. The RTP clients are connected by an Ethernet LAN, which is divided into two independent networks, network A and network B, by the multi-RTCP device as shown in Figure 8.

Network A is simulated by a network simulator which provides jitter on the assumption of a wireless link with a retransmit mechanism. The other network is a bandwidth-restricted network which simulates a congested fixed network (Table 1). Sixteen connections of RTP/RTCP packets generated simultaneously traverse the two networks and are monitored at the interconnection point by the Multi-RTCP device.

The RTP clients send the RTP packets, which carry repeated 16 sec sample voices defined in the ITU-T Rec. P.50 [9] for evaluation. The receivers of the RTP packets record these sample voices for 512 sec and slice them each into 16 sec samples in order to evaluate the speech quality using the ITU-T Rec. P.862, Perceptual Evaluation of Speech Quality (PESQ) [10]. PESQ is designed to predict subjective mean opinion scores (MOS) of degraded speech samples. PESQ scores range from -0.5 to 4.5, with higher scores indicating better quality. We choose the AMR (Adaptive Multi-Rate) codec, which is standardized by the European Telecommunications Standards Institute (ETSI) and is a standard in 3GPP and 3GPP2 [11, 12]. AMR is a variable bit rate codec that operates a bit rate in the range of 4.75 kbit/s to 12.2 kbit/s.

Table 2 shows an example of the PESQ values of the AMR codec. The quality of the higher bit rate of the AMR codec is clearly better than that of the lower one.

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The transmission mechanism based on the Multi-RTCP scheme is able to maintain the appropriate bit rate of the AMR codec for improved end-to-end speech quality through consideration of each portion of the simulated IP-based FMC networks. We evaluated the speech quality under the previously- mentioned condition.

4.1 Evaluation of the packet loss rate

The relation between the congestion level and packet loss rate is shown in Figure 9. The figure contrasts the transmission control mechanism using the RTT information of the divided network B, (Multi-RTCP Scheme) and the entire network A and B (Existing Scheme).

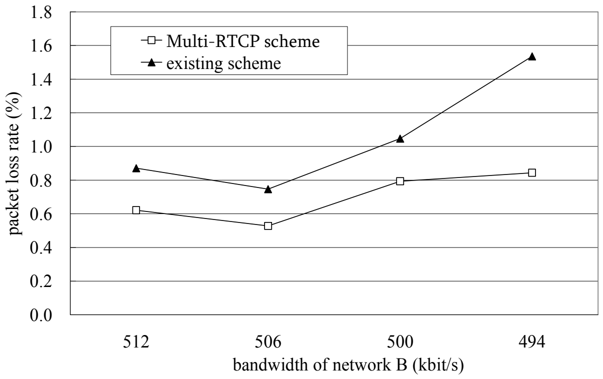

Figure 9. Congestion level versus packet loss rate

The figure indicates that the Multi-RTCP scheme is highly superior in cases where the congestion in the fixed network is heavy. This result means that RTP clients which use the end-user QoE and the QoS metrics of the congested network can adjust the sending bit rate appropriately. However, it is difficult for RTP clients that use the end-user QoE and the QoS metrics of the entire network to adjust the sending bit rate, and it is easy for the congested network to drop the jittered packets.

The transmission control mechanism applied to the evaluation is based on the RTT in RTCP messages. When the Multi-RTCP scheme is available, RTP clients adjust the transmission bit rate based on the RTT value in extra RTCP packets. Under these circumstances, only the RTT from congestion is used and RTT from network infrastructure is ignored. When the congestion increases the RTT and crosses the threshold, the bit rate is decreased. Conversely, when the congestion issue is resolved, the bit rate will increase. Using the original RTCP environment as a comparison, the transmission control mechanism is based on the RTT in RTCP packets, or the RTT of the entire network. In this environment, the RTT and threshold for adjusting the transmission bit rate are the sum of the entire network.

In fixed networks, the main reason for the increase in interarrival jitter and fractional packet loss in RTCP RR is congestion. Therefore, when RTP clients know that interarrival jitter and fractional packet loss are increasing, compression of the transmission bit rate is effective for solving the congestion. In contrast to fixed networks, the quality of mobile networks depends on that of wireless links. When poor quality of wireless links causes high interarrival jitter, the reduction of the sending bit rate of the speech codec is not always valid.

4.2 Evaluation of objective speech quality

Figure 10 shows the change in average PESQ values of all of the 16 RTP streams when the bandwidth of network B is 494kbit/s. The sending bit rate of the AMR codec is

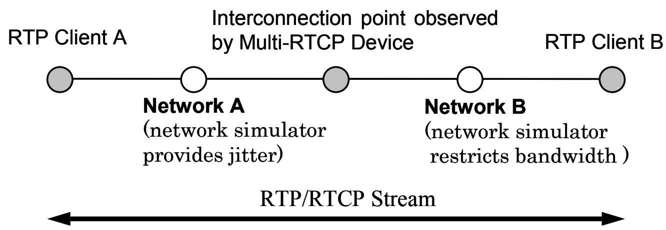

Figure 8. Configuration of the evaluation environment

Table 1. Configuration of networks for experiments

	Network A	Network B
Jitter	Average: 80 msec Standard deviation: 20 msec	No jitter
Bandwidth	No restriction	494 kbit/s, 500 kbit/s, 506 kbit/s, 512 kbit/s
Queue	Unlimited	4 Kbytes
Packet loss	No packet loss	No packet loss
Assumed	Wireless network	Congested fixed network

Table 2. Example of the PESQ values for AMR codec

Bit rate	PESQ
4.75 kbit/s	3.18
12.2 kbit/s	3.50

adjusted depending on the RTT of network B in the extra RTCP packets or on the entire networks in the RTCP packets. In the former, the RTT of network A is just ignored. As time passes, the sending bit rate of the former is suitably adjusted and the speech quality is maintained. In the latter, there is trembling. This means the RTT information of entire network A and B is not sufficient to understand the reason for the fluctuation of RTT, the jitter of network A or the congestion of network B. Especially, RTP clients mistake the large jitter for the delay from the congestion and make wrong adjustments, which causes an unstable degradation in speech quality.

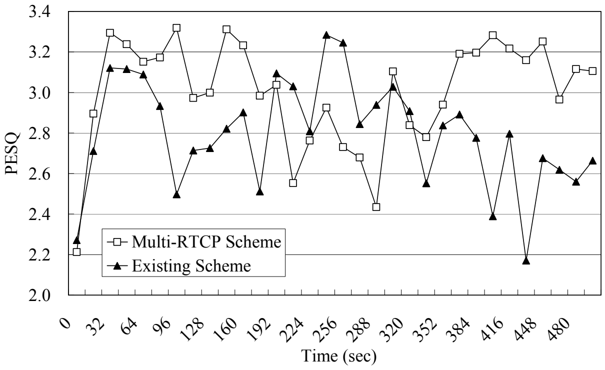

Figure 10. Evaluation of PESQ values

5. Network resource management architecture for RACF

In this section we provide a network resource management system for RACF [3] that can be operated based on the end-user QoE and the network QoS metrics information provided by the network node based on our proposed scheme mentioned in Section 3. The configuration of functional entities is shown in Figure 11. In the figure, the Policy Enforcement Functional Entity (PE-FE) in the network node provides a function of the rate limiting and bandwidth allocation according to an instruction of RACF.

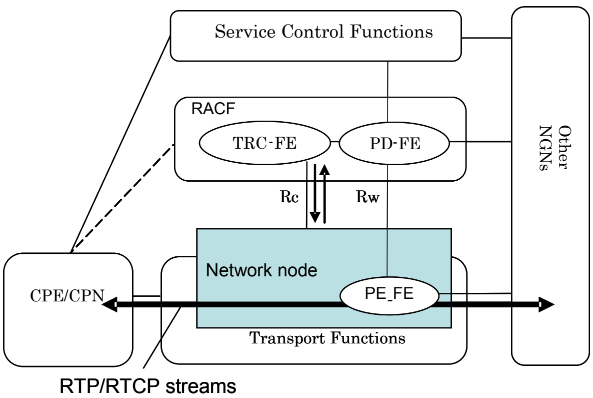

Figure 11. Configuration of functional entities

5.1 SOA-based network resource management

Unified control of a multi-layer network is a motivation for using control plane techniques, while its operation should be co-coordinated with management systems, such as operation systems (OSs) and service provisioning systems, in order to satisfy service requirements. Service oriented architecture (SOA) is a design concept of harmonizing systems using a Web services interface (WSI). SOA is also being adopted for the interface specification between OSs [13]. A concept of Network Resource Management system (NetRM) has been developed to provide service-oriented network services via WSI [14]. Based on the service requests, NetRM has the role of taking decisions to allocate and control the appropriate end-to-end network resources to meet the service requirements, such as bandwidth, end-to-end latency and requested time considering the end-user QoE and the QoS metrics provided by the proposed scheme in Section 3. Through the network service WSI, NetRM provides on-demand network services with in-advance reservation capability. NetRM includes resource control functions related to Policy Decision Functional Entity (PD-FE) and Transport Resource Control Functional Entity (TRC-FE) in RACF.

5.2 NetRM functional architecture

Figure 12 shows the network resource management architecture with a functional block diagram of NetRM. Using an SOA bus where each system is connected as a Web service described by WSI, a provisioning system cooperates with NetRM and other OSs to discover information and to reserve resources required for guaranteed or scheduled services. NetRM provides guaranteed and scheduled network services to network service clients via the SOA bus.

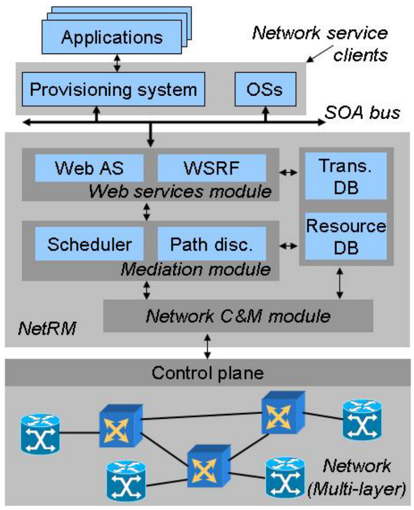

Figure 12. Network resource management architecture

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A Web service module of NetRM handles service messaging using a Web application server (Web AS). Implementation based on the Web service resource framework (WSRF) established by the OASIS allows NetRM to support stateful request handling and avoidance of unexpected reservation resource dissipation. Following a request processed by a Web service module, a mediation module assigns appropriate resources using optimized-path discovery and scheduling functions. A mediation module allows both on-demand and in-advance network services. The information about reservations and network resources is dynamically managed by a transaction and a resource database, respectively. A network control and management (C&M) module interworks with the control plane (e.g. MPLS, GMPLS) of the physical network and utilizes its distributed operation for multi-layer provisioning and synchronization of the network state.

6. IP-based FMC services

As an example of the IP-based FMC service, the user will be able to choose an optimal access network according to the location. In this service, the user can enjoy a telephony service in the home through a WLAN that is connected to the fixed broadband access network by using a dual mode mobile phone. When the user has to go out, the user can switch to the mobile access network while seamlessly continuing the communication. Furthermore, not only fixed/mobile phone but many types of devices will be able to be used in NGN, like PC, PDA and other information appliances. In addition, the services in NGN are not limited to voice telephony but also include text messaging, video telephony, conferencing and any other type of applications. This means that a user will have many types of devices and applications. Following is an example scenario and functionalities for the service session mobility over IP-based FMC networks considering the end-user QoE.

6.1 Usage case scenario

In this section, we explain a usage case scenario of service mobility. Now, Alice and Bob are communicating with video telephony service on their PCs at their offices. Although closing time comes, they still have some business to discuss, so he switches to voice telephony on his mobile phone and continues communication on his way home. After he arrives back at his home, he again switches to video telephony using TV as a video device and his mobile phone as an audio device, and finishes their discussion.

6.2 Service mobility

From the above usage case scenario, we abstract three essential user level resources that compose the communication service. In order to adapt the communication service to the user's demand, situations and preferences, the resources should be able to be combined and switched seamlessly. The three resources are application, access network and device.

By application switching, the user can use an appropriate application regarding the current situation or demand. Switching the access network allows the user to use an efficient access network in terms of cost or performance at the user's current location. Device switching also allows the user to communicate with the appropriate device in the current situation. In addition, there is also a composite case that switches two or three kinds of resources at once. For example, both application switching and device switching are performed to switch an application which is not supported on the current user device.

6.3 Session control for service mobility

In order to achieve the above types of switching, we developed a session control function based on SIP [15]. By using this method, sessions between a caller and a callee are established and disconnected according to the resources they agree to use. The negotiation of the resources to be used is done in end-to-end fashion before the actual session control is performed. For this purpose, two kinds of list are exchanged between the users. One is a list of all the resources that a user has. From this list, a user can know possible applications (name of the application, required media type, direction) and corresponding end points (IP address, port number) to be connected to use the applications. Because a user may have multiple devices, the list will contain all the applications that the devices support, and the end points will be varied according to the devices and access networks they apply to. The other is a list of the resources that are willing to be used by a user. When a user receives the list from its corresponding user, the user gives acceptable resources according to the situation, preferences, etc. After negotiation is complete, a session control is performed to establish required sessions for new applications and new end points, and to disconnect unnecessary sessions of previous applications and end points.

6.4 QoE in FMC Service

FMC is network agnostic by its definition, and the quality of communication services is affected by the characteristics of the underlying network. To take service mobility as an example, if a user switches the current application from voice to video telephony regardless of using a narrow access network that does not have any control function for network QoS metrics, the quality of the application may be degraded.

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From the viewpoint of the FMC services mentioned above, we believe that a service control scheme considering end-user QoE is imperative. In order to maintain QoE of the communication service, the network resource management function mentioned in Section 5 should be supported in each access network and it should be utilized when the communication service is going to be set up to allocate required bandwidth or offer a certain service class.

Furthermore, if the current characteristic of each access network like available bandwidth or delay can be known from the network resource management function, it will help make a decision as to which access network should be applied to by an application in the face of service mobility.

7. Conclusions

The challenges and requirements of service control for supporting end-user QoE and QoS metrics in converged NGNs are crucial issues. Considering the wide range of fluctuation of QoS metrics in IP-based FMC networks, in this paper, a new service control scheme based on the end-user QoE predicted by per network-segment based QoS metrics using RTP/RTCP is proposed. The proposed scheme is a wide-ranging scheme, because RTP and RTCP are commonly used protocols for transmitting multimedia streams. Furthermore the proposed prediction method for end-user QoE provides more or less the same accuracy by comparison with similar existing methods that require much higher computational cost. The results for a typical service control, which is the bit rate control of a speech codec, show that the proposed scheme works well to maintain speech quality even where the transmission throughput fluctuates greatly in FMC networks. The relevant study issue based on the proposed scheme is a current study item of ITU-T Q4/13 which discusses requirements and framework for QoS for NGN. Based on the proposed scheme, the resource status information such as bandwidth is collected in order to realize RACF controls.

The performance of the proposed scheme can be improved by adding the capability to make notification of the handover parameters inherent in each type of wireless access media. In this respect, we also discussed preliminary network resource management architecture considering a new type of promising IP-based FMC service that realizes the proposed QoE-aware session control for service mobility.

Furthermore, in the future, it is easy to imagine that one CPE will handle multiple types of service media, such as voice, video, text, etc., while communicating with one or more CPEs at the same time. The proposed service control scheme using RTP/RTCP can be applied to control each service media stream independently while considering the end-user QoE in such environments of concurrent multimedia services in the NGN era.

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