SIP Method Escalation Algorithm for NGN Application Network Interface

Kiyoshi UEDA†*, Hitoshi SHIMIZU†, Tetsuya IWATA†, Hiroshi SUNAGA††*, and Akira FUKUDA†††
Member, IEEE

***Abstract*—The Next-Generation Network (NGN) has Application Network Interfaces (ANIs) by which 3rd party application services are easily created. For session-control ANI (for example, "makeCallSession" of the Parlay-X), Application Server (AS) functions process a call using three SIP methods (REFER method, 3rd Party Call Control (3PCC) method, Media Bridge method). We propose a SIP method escalation algorithm where each escalation level corresponds to the SIP protocol capabilities a terminal has. We evaluate the proposed algorithm from the viewpoints of resource consumption and session connecting delay, and performed research on issues regarding commercial use. The results are expected to contribute to the future Service Delivery Platform (SDP) or Proactive network Provider Participation for P2P (P4P).**

***Index Terms*— NGN ANI, Parlay-X API, SIP**

I. Introduction

An efficient, secure, and flexible Next-Generation Network (NGN) enables the provision of seamless IP-based services by using optical access and broadband wireless access. The NGN will have specific functions for controlling traffic during times of congestion and for restricting unauthorized traffic, as well as for handling cyber-terrorism and physical damage due to natural disasters. By ensuring communication quality and bandwidth appropriate for the various types of services, the NGN will provide safe, secure, and convenient services that combine the advantages of the existing telephone network and the IP network.

The NGN adopts a layered structure model so that it can respond better to technological advances and service diversification. The NGN provides interfaces between the network and application management systems. The interfaces enable application service providers (ASPs), video distributors, and others to make use of the NGN. Then, ASPs use the NGN interfaces and provide a wide range of application services. The service control functions of the NGN conform to the IP multimedia subsystem (IMS) [1], which is compatible with the layered structure model being standardized by the International Telecommunication Union (ITU) (Fig. 1) [2].

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†NTT Network Service Systems Laboratories, ††NTT Network Innovation Laboratories, NTT Corporation, 9-11 Midori-cho 3-Chome Musashino-Shi, Tokyo 180-8585 Japan (Kiyoshi Ueda phone: +81-422-59-3564; e-mail: ueda.kiyoshi@lab.ntt.co.jp).

†††Kyushu University, 744 Motooka, Nishi-ku Fukuoka 819-0395, Japan

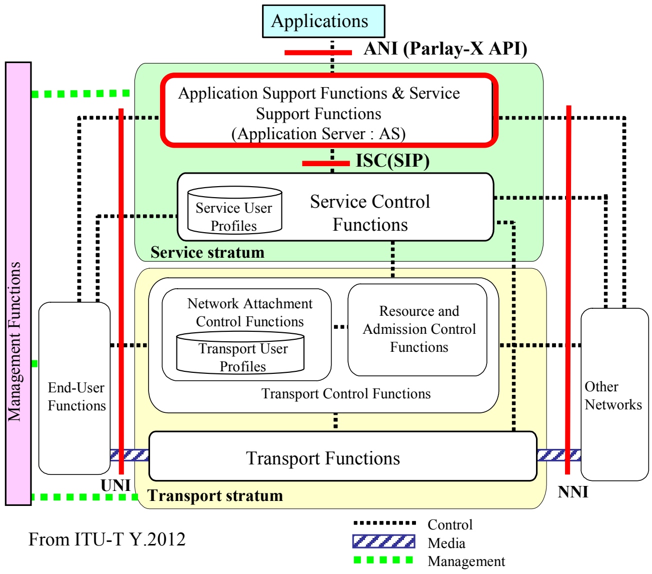

Fig. 1. Architecture of NGN (ITU-T Y.2012).

Application Support Functions and Service Support Functions (ASFs and SSFs) are added to the NGN architecture, which provide an Application Network Interfaces (ANIs). Third-party application services are easily created using ANIs (e.g., Parlay-X Web Service APIs [3] [4]). The ASF and SSF serve as an interworking entity between the ANI and ISC (SIP) [5] in the Service stratum. We introduce Application Server (AS) functions, which process a call for the ANIs (Parlay-X APIs) using SIP methods.

There are some SIP methods for sessions between terminals in the NGN. The most efficient SIP method, with respect to SIP protocol capabilities that a terminal has, is different from the viewpoint of resource consumption. We proposed a SIP method escalation algorithm considering the heterogeneous environment constructed by various networks and terminals. We evaluated the resource consumption of the proposed algorithm and performed research on session-connecting delay characteristics and issues regarding commercial use.

The remainder of this paper is organized as follows. We introduce a service example using ANIs. Section 3 describes the session control function using the parlay-X APIs. Section 4 describes the proposed algorithm that escalates SIP methods for the parlay-X APIs. Section 5 presents the results of the evaluation. Section 6 concludes with a brief summary and mentions topics for further study.

II. Service Example Using ANI

The proposed algorithm is expected to contribute to the future Service Delivery Platform (SDP) [6] [7] or Proactive network Provider Participation for P2P (P4P) [8] [9].

SDP is attractive for making services that mash up various services more widespread by using such a network control function (via ANIs) (see Open Mobile Alliance (OMA) [10]).

The Distributed Computing Industry Association and its "P4P Working Group" is working on a new peer-to-peer protocol, called P4P. It is described as a "carrier-grade peer-to-peer file transfer system." The P4P protocol aims to reduce traffic by using network topology data to select peers intelligently, instead of at random, thus increasing routing efficiency. This is obvious: when the traffic has to take more routers or hops, it is less efficient. A P2P network is an overlay network constructed of terminals that are connected through a physical IP network. The topology of the P2P network above the NGN is configured by SIP sessions between the terminals, which is provided by the service stratum.

Using session-control ANIs, we can build an efficient P2P network not only for constructing a P2P network, but also for reconfiguring a P2P network for adapting to traffic duration. The number of useless relay-forwarding packets can be reduced by switching peer to peer sessions using ANIs. As shown in Fig. 2, when peer C becomes congested, peer C evaluates the relay-forwarding packet density of each source peer and destination peer combination. If relay forwarding between peers A and B was most congested, peer C would send a "makeCallSession" to ANI for connecting between peer A and peer B.

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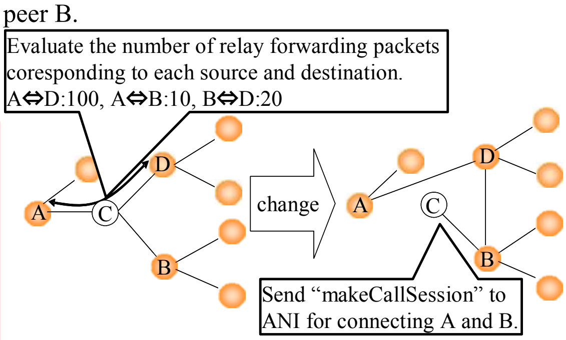

Fig. 2. Reconfiguring P2P network adapting to traffic duration.

III. Session Control Function Using Parlay-X APIs

A. Session control function

The Parlay-X 3.0 API specification has sufficient capability to control the call session for the ANI as follows (Fig. 3).

(1) Applications request to make a session between terminals A and B by "MakeCallSession"

(2) The AS responds with "CallSessionID"

(3) The AS calls terminal A

((3)' When the AS continuously controls and switches sessions with terminal A (session multiple switching is necessary), the AS calls another terminal, for example, a media server for stream guidance)

(4) The AS calls terminal B

(5) Terminal A starts a media stream with terminal B

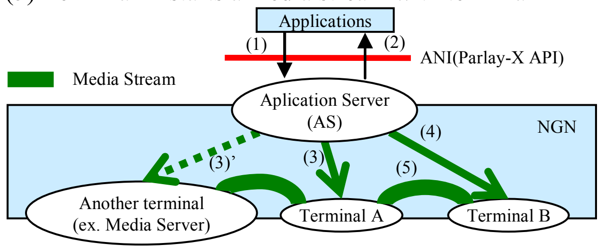

Fig. 3. Session control function.

B. Three SIP methods for "makeCallSession"

There are three SIP methods (Fig. 4) for controlling the call session ("makeCallSession" API).

1) REFER method (RFC 3515 [11])

The AS sends a REFER request to terminal A, which orders terminal A to send an INVITE request to terminal B.

2) 3rd Party Call Control (3PCC) method (RFC 3725 [12])

The AS sends INVITE requests to terminal A and terminal B; then, the AS sends an UPDATE request to terminal A and terminal B, which orders each terminal to make a media stream with each other.

3) Media Bridge method

The AS makes a session with terminal A and establishes a media stream between media transport equipment and terminal A. Similarly, the AS makes a session with terminal B and establishes a media stream between the media transport equipment and terminal B. Then, the AS (media transport equipment) bridges these media streams.

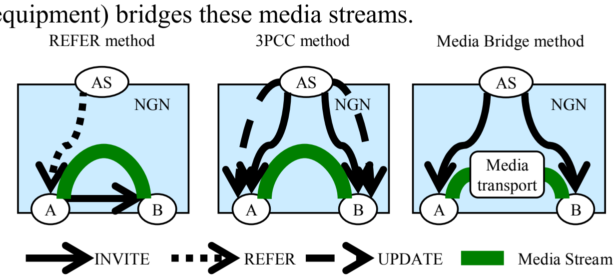

Fig. 4. Three SIP methods for "makeCallSession".

C. Characteristics of SIP methods

1) The REFER method

Resource consumption is low, but the session between terminals A and B is uncontrollable (multiple switching in a session is impossible). The terminals must have REFER method capability.

2) The 3PCC method

The AS can control the session between terminals A and B (session multiple switching is possible), but the terminals must have UPDATE method capability.

3) The media bridge method

The AS can control the session between terminals A and B (session multiple switching is possible), and all terminals are available. Only a limited number of terminals can use the media bridge method because media transport resources are so expensive. This is unsuitable for a wide area network or a metropolitan area network.

TABLE I
THE CHARACTERISTICS OF SIP METHODS.

		REFER method	3PCC method	Media Bridge method
Session control (session switching) capability		Only once switching	Multiple switching	Multiple switching
Resource consumption of lower layer	C-Plane (Service Control Functions)	1 session	2 sessions	2 sessions
	U-Plane (Transport Functions)	1 media stream	1 media streams	2 media streams
AS resource consumption	C-Plane	0 session (AS keeps no session)	2 sessions	2 sessions
	U-Plane (Media Bridge)	0 media bridge	0 media bridge	2 media bridge
Connecting delay (The number of SIP sequences for session establishment or modification)		1 sequence (1 INVITE)	3 sequence (2 INVITEs+ 1 UPDATE)	2 sequence (2 INVITEs)
Terminal capability		One terminal should have a function for REFER	Both terminals should have a function for UPDATE	All terminals are available

IV. PROPOSED ALGORITHM

For services that need multiple switching in a session, AS cannot use the REFER method. If terminals have REFER method capability but do not have UPDATE method capability, the terminals must not subscribe to services that need multiple switching in a session. Hence, assuming that multiple switching in a session is unnecessary, we propose a SIP method escalation algorithm that checks the terminal’s protocol capability (Fig. 5). Using this algorithm, network session control ANI becomes more practical because all terminals are widely available and resource consumption is kept low.

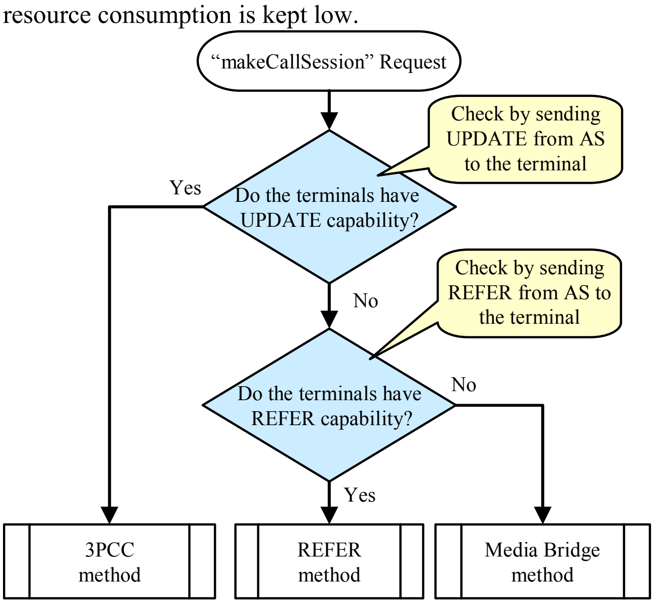

Fig. 5. Proposed algorithm.

Step 1: The AS sends UPDATE to terminals and checks responses to determine whether the terminal has UPDATE capability. If the terminal has UPDATE capability, the AS selects the 3PCC method.

Step 2: If the terminals do not have UPDATE capability, the AS sends the REFER to the terminals and checks the response to determine whether the terminal has REFER capability. If the terminal has REFER capability, the AS selects the REFER method.

Step 3: If the terminals do not have UPDATE and REFER capability, the AS selects the Media Bridge method.

If multiple switching in a session is necessary, not all terminals can use the REFER method, so the proposed algorithm must be changed so that step 2 will be skipped.

V. EVALUATION

We evaluated the proposed algorithm by comparing with a simple algorithm that uses only the media bridge method with respect to resource consumption and session connecting delay.

A. *Resource consumption evaluation*

1) *Resource consumption model*

We designed a resource consumption model to evaluate our proposed algorithm. The total resource consumption of method j (C_j) is calculated as follows.

$$C_j = \sum_i \{w_i N_{ij}\} \qquad (1)$$

w_i: Resource consumption ratio of resource i
N_{ij}: The amount of resource i consumed for method j

TABLE II
RESOURCE CONSUMPTION OF THE METHOD J.

Method j / Resource i			REFER method	3PCC method	Media Bridge method
Resource consumption ratio (W_i)			The number of resource consumption for the method j (N_{ij})		
Resource consumption of lower layer	C-Plane	w_1	1 session	2 sessions	2 sessions
	U-Plane	w_2	1 media stream	1 media stream	2 media streams
AS resource consumption	C-Plane	w_3	0 session	2 sessions	2 sessions
	U-Plane	w_4	0 media bridge	0 media bridge	2 media bridges
Total resource consumption ratio of method j: C_j			w_1+w_2	$2w_1+w_2+2w_3$	$2(w_1+w_2+w_3+w_4)$

2) *Resource consumption ratio*

We estimated the ratio of the resource consumption (w_1, w_2, w_3, w_4) as follows.

The correlation between “Resource consumption of lower layer” (w_1, w_2) and “AS resource consumption” (w_3, w_4) is estimated to be the same as the correlation between telephone charge and audio guidance telephone charge because the communication charge represents the network costs.

The correlation between “AS resource consumption C-Plane” (w_3) and “AS resource consumption U-Plane” (w_4) is estimated to be the same as the correlation between SIP session management equipment price and media transport equipment price because the AS consists of SIP session management equipment and media transport equipment.

The correlation between “Resource consumption of lower layer C-Plane” (w_1) and “Resource consumption of lower layer U-Plane”(w_2) is estimated to be the same as the correlation between “AS resource consumption C-Plane” (w_3) and “AS resource consumption U-Plane” (w_4).

TABLE III
RESOURCE CONSUMPTION RATIO.

Resource j		Resource consumption ratio (w_i)	Ratio of SIP session management equipment price to media transport equipment price	Ratio of telephone charge to audio guidance telephone charge
Resource consumption of lower layer	C-Plane	0.58	15%	¥3.86/3min
	U-Plane	3.28	85%	
AS resource consumption	C-Plane	0.65	15%	¥4.33/3min
	U-Plane	3.68	85%	

	REFER method	3PCC method	Media Bridge method
Total resource consumption ratio of method j:C_j	3.86	5.74	16.37

3) Total Resource consumption evaluation of proposed algorithm and simple algorithm

Total resource consumption (C) is the sum of the resource consumption of method j (C_j) multiplied by the probability of each method j (P_j).

$$C = \sum_j P_j C_j \qquad (2)$$

For the proposed algorithm, we present the probability of each method j (P_j) using the following parameters (TABLE IV).

-The probability that terminals have REFER capability:
P_{REF} ($0 < P_{REF} < 1$)

-The probability that terminals have UPDATE capability:
P_{UP} ($0 < P_{UP} < 1$)

If multiple switching in a session is necessary, the algorithm will be changed and the probability becomes simple.

For a comparative study, we propose a simple algorithm, which uses only a media bridge method. In this case, the probability of the terminal (P_{simp}) is always 1.

TABLE IV
PROBABILITY OF METHOD J.

Proposed algorithm

Method j	Probability of method j (P_j)
REFER method	$(1-P_{UP}^2)P_{REF}$
3PCC method	P_{UP}^2
Media Bridge method	$(1-P_{UP}^2)(1-P_{REF})$

If session multiple switching is necessary

Method j	Probability of method j (P_j)
3PCC method	P_{UP}
Media Bridge method	$(1-P_{UP})$

Simple algorithm

Method	Probability (P_{simp})
Media Bridge method	1

Depending on the terminal’s SIP protocol capability, the total resource consumption of the proposed algorithm compared with the simple algorithm (media bridge method only) is shown in Fig. 6. The special case if multiple switching in a session is necessary is shown in Fig. 7. In the figure, the graph presents the total resource consumption of the proposed algorithm, and the resource consumption of the simple algorithm is at the bottom plane.

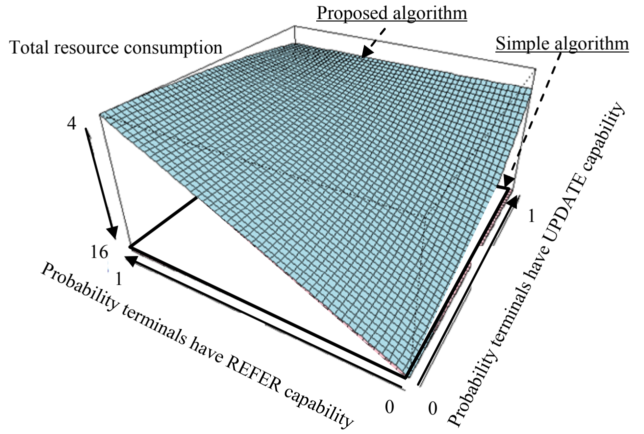

Fig. 6. Total resource consumption.

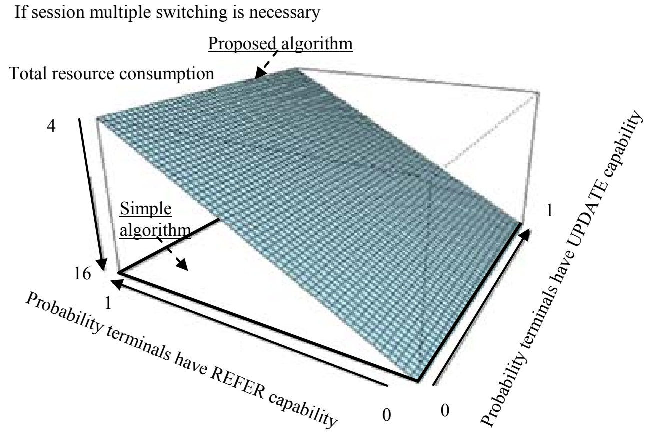

Fig. 7. Total resource consumption if session multiple switching is necessary.

If the probability that terminals have REFER capability or the probability that terminals have UPDATE capability becomes bigger, the total resource consumption becomes smaller. Even if multiple switching in a session is necessary, the total resource consumption is smaller than the simple algorithm.

In all cases, the proposed algorithm is efficient from the viewpoint of resource consumption.

B. Session connecting delay evaluation

1) Session connecting delay of the methods

We designed the session connecting delay model to evaluate our proposed algorithm and performed research on the subjects for commercial use. Depending on the terminal SIP protocol capability, the number of SIP sequences for session establishment is different. This has an impact on session connecting delay. Total delay (D) is the sum of the delay of method j (D_j) multiplied by the probability of each method j (P_j).

$$D = \sum_j P_j D_j \qquad (3)$$

The probability of each method is shown in TABLE IV.

Session connecting delay ratios of each method are estimated to be the same as the number of SIP sequences of each method for session establishment (TABLE V).

The session connecting delay of the simple algorithm (D_{simp}) is the session connecting delay ratio of the media bridge method, so D_{simp} is always 2.

TABLE V
SESSION CONECTING DELAY OF METHOD J.

Method j	REFER method	3PCC method	Media Bridge method
Session connecting delay ratio of method j (D_j) (The number of SIP sequences for session establishment)	1 (1 INVITE)	3 (2 INVITEs+ 1 UPDATE)	2 (2 INVITEs)

2) Session connecting delay evaluation of proposed algorithm and simple algorithm

A comparison of the session connecting delay of the simple algorithm (media bridge method only), and the proposed algorithm can be seen in Fig. 8. The special case if multiple session switching is necessary is shown in Fig. 9. In the figures, the graph presents the connecting delay of the proposed algorithm, and delay of the simple algorithm is presented at the plane in which delay equals 2.

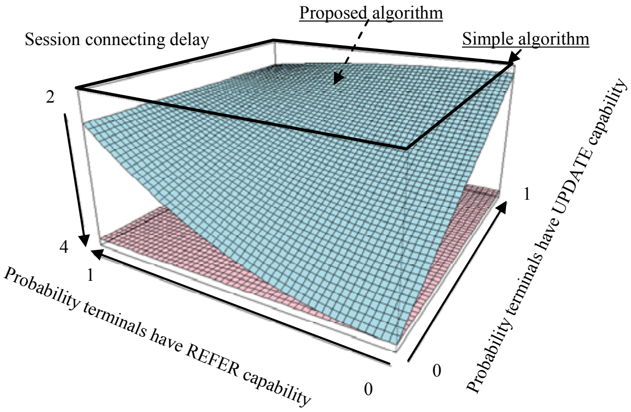

Fig. 8. Session connecting delay.

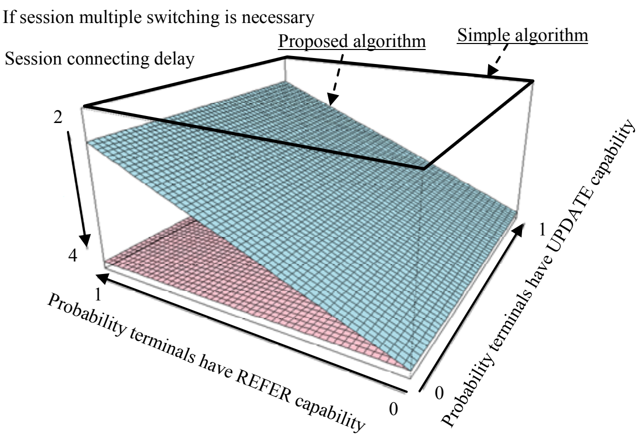

Fig. 9. Session connecting delay if session multiple switching is necessary.

If the probability that terminals have REFER capability becomes bigger, session connecting delay becomes shorter. If the probability that terminals have UPDATE capability becomes bigger, the proposed algorithm needs more SIP sequences.

In some cases, the proposed algorithm needs more SIP sequences compared with the simple algorithm. For introducing the proposed algorithm in commercial services, we should consider the influence of the probability if the session connecting delay becomes longer.

VI. CONCLUSION

In this paper, we proposed a function between Parlay-X API and SIP processing, which has a SIP method escalation algorithm corresponding to the terminal's SIP protocol capability.

We evaluated the proposed algorithm from the viewpoint of network resource consumption and session connecting delay compared with the simple algorithm, which uses only the media bridge method.

We confirmed that resource consumption of the proposed algorithm is low compared with that of the simple algorithm. The proposed algorithm needs more SIP sequences compared with the simple algorithm, so we must be careful to introduce the proposed algorithm for commercial services.

We studied the algorithm for the "makeCallSession" Parlay-X API, and we will study the ASF and SSF functions of other APIs and services (for example, call transfer service using the "handleCalledNumber" API). We evaluated these algorithms from the viewpoint of the terminal's static ability. We will consider the dynamic alteration of a terminal's capability for various rich services. We will improve the proposed algorithm and evaluate the aspect of terminal ability duration in a future study.

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