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Media Gateway Control Protocol (MGCP)
Call Flow Test Case 1

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1. Status of this document

Abstract

The IETF Megaco working group is currently working on defining a protocol for controlling Media Gateways (MG) from external control elements such as a Media Gateway Controller (MGC). Different models have been proposed to address this problem, and in order to judge the different models, a series of test cases will be considered.

This document explains how the Media Gateway Control Protocol (MGCP) can be used to handle the first test case, which considers a Voice over IP gateway with directly subtending DTMF lines placing a call to an H.323 client. The document contains an introduction with further details on the test case, two different call flows to solve it, followed by the conclusion of the test case. MGCP is defined in a companion document [0].

2. Introduction

The Media Gateway Control Protocol can be used to control different types of endpoints, such as "residential gateways", "trunking gateways" or "packet relays". This document will show how MGCP can be used to control a residential gateway in the Megaco call flow test case 1, which considers a Voice over IP gateway with directly subtending DTMF lines placing a call to an H.323 client. We first present the Megaco test case 1, and we then present an example call flow for solving it. Following that, we discuss possible optimizations to the call flow, and we then present the conclusions reached in using MGCP to satisfy the test case.

3. Call Flow Test Case 1 - DTMF line dialing into VoIP Gateway

In this section we first present the problem statement for test case 1, and we then present the assumptions for the test case that were made.

3.1. Problem Statement

In this section, we present the problem statement given for the Megaco call flow test case 1. The problem is formulated as follows:

Setting: a VoIP Gateway provides service to directly subtending DTMF lines.The standard North American dialling plan is in use: seven digits beginning with 2-9 for a local number, 10 digits for a North American toll number, the single digit 0 for an operator, 411 for directory information, and 911 for emergency service. Overseas dialling is 011 followed by a variable number of digits. The subscriber actually dials a seven-digit local number which has been assigned to a friend’s H.323 client installation. The Gateway consults with an H.323 Gatekeeper to resolve the dialled digits into the H.323 client’s address, then sets up an audio call to that client. At the end of the call the MGC component of the Gateway is required to report to a billing system for billing purposes the following items:

-- the time the call began (subscriber off-hook recorded)

-- the time media transfer between the calling and called parties began

-- the time media transfer between the calling and called parties ended

-- total media packets and octets respectively transmitted by the Gateway to the called party

-- total media packets and octets respectively received by the Gateway from the called party

-- difference between total media packets actually received at the Gateway and the total number expected

-- mean interarrival jitter as calculated at the Gateway for received packets. (Notice I don't trust the H.323 client to report anything that can be used for billing.)

Call flow:

```
        Calling          Gateway                Called      Gatekeeper
           |  1. Off-hook -->|                   |                 |
           |<-- 2. Dial tone |                   |                 |
           |3. First digit-->|                   |                 |
           |<4. No dial tone |                   |                 |
           |5. Digit      -->|                   |                 |
            ...
           |10. Digit     -->|                   |                 |
           |                 |    11. Admission Request (ARQ)----> |
           |                 |<---12. Admission confirm (ACF)      |
           |                 |    13. SETUP   -->|                 |
           |                 |                   |    14. ARQ  --> |
           |                 |                   |<-- 15. ACF      |
           |                 |<-- 16. ALERTING   |                 |
           |<-- 17. ringback |                   |                 |
           |                 |<-- 18. CONNECT    |                 |
           |<- 19. cutthrough|                   |                 |
           ...
           |  20. On-hook -->|                   |                 |
           |                 |    21. RELEASE -->|                 |
           |                 |<-- 22. RLC        |                 |
           ...
```

Notes: (numbered by the steps to which they apply)

11. Admission Request contains dialled number as an E.164 address. Also requests total bandwidth of 128 kbs (64 kbs in each direction to accommodate G.711 coded audio).

12. Admission Confirm provides call signalling address and port for the called H.323 client. It indicates that only 80 kbs is available.

13. To meet the bandwidth restriction, the SETUP message contains a fastConnect element which offers to send G.711 mu-law audio and receive G.729 Annex C audio or vice versa. It supplies port addresses at which it will receive RTP (incoming media) and RTCP (for the media it transmits). The SETUP message also contains a

Boolean indicating that the called H.323 client should not transmit any media packets until it issues a CONNECT message.

14-15.
The called H.323 client gets permission for its intended bandwidth usage.

16. The called H.323 indicates that it is alerting the called party, and that it has chosen to receive G.711 mu-law and send G.729 Annex C. It supplies the port at which it will receive the G.711 and the port at which it will receive RTCP corresponding to the G.729 it sends out.

17. It is assumed for this scenario that the Gateway is responsible for supplying ringback tone to the caller.

18. The CONNECT message from the called H.323 client indicates that the called party has answered and the call has entered talking state.

19. The Gateway must discontinue ringback tone and cut through talking in both directions.

20. Assumed for this scenario that the caller is the first to hang up.

The Disengage sequences which must pass between the Gatekeeper and the Other two H.323 entities at the end of the call are omitted, since they are not relevant.

Assumptions

* The description of the dialing plan does not include any restrictions on the structure of the 10-digit number, and it is therefore assumed that the 10-digit number may start with any digit, incl. 2-9.

* We assume that international numbers (excluding direct-dial prefix) may consist of as few as 1 digit.

* We will be using H.323 version 2.

4. Call Flows

We present two different call flows to test case 1. The first call flow illustrates how MGCP can be used to satisfy the test case when two separate connections are used. The second call flow illustrates an alternative solution where we only use one connection.

4.1. Two Connection Call Flow

The diagram below illustrates the basic call flow:

Step	User	MG	MGC	H.323 EP	GK
0		<-	RQNT		
		Ack	->		
		(ready)			
1	Off-	NTFY	->		
1a	hook		(off-hook		
			recorded)		
		<-	Ack		
2	(dial	<-	RQNT		
	tone)	Ack	->		
3	digit				
4	(no dial				
	tone)				
5	digit				
	...				
10	digit				
10a	(match)	NTFY	->		
		<-	Ack		
10b		<-	RQNT+		
			CRCX-1		
		Ack(SDP-1)	->		
11			ARQ	- -	->
12			<-	- -	ACF
12a		<-	CRCX-2		
12b		Ack(SDP-2)	->		
13			SETUP	->	
14				ARQ	->
15				<-	ACF
16			<-	ALERTING	
17	(ring	<-	RQNT +		
	back)		MDFY-1(SDP),		
			MDFY-2(SDP)		
		Ack, Ack	->		
18			<-	CONNECT	
19	(cut-	<-	RQNT		
	through)		+MDFY-2		
		Ack	->		
			(full-duplex		
			media transfer		
			recorded)		

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