

Unity Connection TIMG Does Not Route Calls Correctly

Contents

[Introduction](#)

[Problem](#)

[Solution](#)

[Related Information](#)

Introduction

This document describes the issue when calls that come in to Slave T1 IP Media Gateway (TIMG) or PBX IP Media Gateway (PIMG) are not routed correctly. TIMGs and PIMGs make it possible for PBXs to integrate to Unity Connection for voicemail access. Some PBXs require that this integration be via Simplified Message Desk Interface (SMDI), MCI, or MD-110. This means that calling information will be passed via a serial port connection from the PBX to the TIMG or PIMG. The TIMG or PIMG that the serial cable connects to will be configured as a Master. If there are other TIMGs or PIMGs required, these will be configured as Slaves and will look to the Master for calling information.

Problem

There are two or more TIMGs/PIMGs with a Master and Slave configuration. When a call comes into the Master, the call is forwarded to the proper Unity Connection voicemail box greeting.

Here is an example screenshot of the page from a Master PIMG:

Config > Serial > Switch Protocol

Status

Summary
Alarms
TDM
VoIP
Serial
Call Log
MIB-II
Statistics

Configuration

Import/Export
IP
Mgmt Protocols
Routing Table
TDM
VoIP
Serial
Tone Detection
Certificates
DSP Settings

Diagnostics

Trace/Logging
Tests

System

Web UI
Password
Upgrade
Restart

Serial Port, COM 1	
* Serial Mode (Master/Slave)	Master
* Serial Interface Protocol	SMDI
MCI Message Extension Length	Six-Digits
MCI Message Type	Type_B
CPID Length	7
Cpid Padding String	
Voice Mail Port Length	2
System Number	1
MWI response timeout (ms)	2000
* IP Address of Serial Server	
Serial Cpid Expiration (ms)	5000

Logical Extension Numbers	
Port #	Port Extension
1	1
2	2
3	3
4	4
5	5
6	6
7	7
8	8
9	9
10	10
11	11
12	12
13	13
14	14
15	15

However, when the call comes into the Slave TIMG the call is answered by the Opening Greeting. The call rolls to the Opening Greeting because the invite sent to Unity Connection from TIMG does not have a 'Diversion:' line within to say which mailbox extension the call should go to.

Here is an example of calling information seen on the Master:

```
08-28 17:54:28.078 [Si      ] Prot    0D
08-28 17:54:28.078 [Si      ] Prot    0A
08-28 17:54:28.078 [Si      ] Prot    4D
08-28 17:54:28.078 [Si      ] Prot    44
08-28 17:54:28.078 [Si      ] Prot    30
```

```

08-28 17:54:28.078 [Si      ] Prot      30
08-28 17:54:28.078 [Si      ] Prot      30
08-28 17:54:28.078 [Si      ] Prot      30
08-28 17:54:28.078 [Si      ] Prot      30
08-28 17:54:28.078 [Si      ] Prot      30
08-28 17:54:28.078 [Si      ] Prot      31
08-28 17:54:28.078 [Si      ] Prot      4E
08-28 17:54:28.078 [Si      ] Prot      31
08-28 17:54:28.078 [Si      ] Prot      39
08-28 17:54:28.078 [Si      ] Prot      31
08-28 17:54:28.078 [Si      ] Prot      38
08-28 17:54:28.078 [Si      ] Prot      20
08-28 17:54:28.078 [Si      ] Prot      39
08-28 17:54:28.078 [Si      ] Prot      31
08-28 17:54:28.078 [Si      ] Prot      39
08-28 17:54:28.078 [Si      ] Prot      33
08-28 17:54:28.078 [Si      ] Prot      33
08-28 17:54:28.078 [Si      ] Prot      33
08-28 17:54:28.078 [Si      ] Prot      33
08-28 17:54:28.078 [Si      ] Prot      34
08-28 17:54:28.078 [Si      ] Prot      38
08-28 17:54:28.078 [Si      ] Prot      35
08-28 17:54:28.078 [Si      ] Prot      20
08-28 17:54:28.078 [Si      ] Prot      0D
08-28 17:54:28.078 [Si      ] Prot      0A
08-28 17:54:28.078 [Si      ] Code      siSrvSerialInputEvent
08-28 17:54:28.078 [Si      ] Prot      From Serial: 0D 0A 4D 44 30 30 30 30 30 30 31
4E 31 39 31 38 20 39 31 39 33 33 33 33 34 38 35 20 0D 0A 19 00
08-28 17:54:28.078 [Si      ] Prot      19
08-28 17:54:28.078 [Si      ] Code      siSrvPrCpidFromSwitch ltn = 1,
src=9133333485, Dst = <NULL>, Redir = 1918, Reason = NoAns
08-28 17:54:28.078 [SiIp    ] Code      sertrans_ServerLocateClient 1
08-28 17:54:28.078 [SiIp    ] Code      sertrans_ServerLocateClient 1=client1
08-28 17:54:28.078 [SiIp    ] Code      _TaskMainClientReceive received data 516
08-28 17:54:28.078 [Si      ] Code      serial_client_cb
08-28 17:54:28.078 [Si      ] Code      SI_TYPE_CPID 1:NoAns (9193333485->->1918)
08-28 17:54:28.078 [Tel-1   ] Code      GetChannelFromLogicalChannelNum
LogicalChanNum 0 span 0 channel 1
08-28 17:54:28.078 [Tel-1   ] Code      tlcasReportNewCpid
08-28 17:54:28.078 [Tel-1   ] Event     Cpid (9193333485,->,->1918,) (NoAns)
08-28 17:54:28.078 [Tel-1   ] Warn      tlcasReportNewCpid err: no call for cpid
08-28 17:54:28.078 [Tel-1   ] Code      tlcasReportNewCpid saving pre-call cpid for
serial
08-28 17:54:29.195 [SiIp    ] Code      _TaskMainServerReceive(4) received 516 bytes
08-28 17:54:29.195 [SiIp    ] Code      _TaskMainServerReceive(4) keep-alive 1
received
08-28 17:54:29.195 [SiIp    ] Code      _TaskMainServerReceive(4) sending keep-alive
response

```

Here is an example of a problem invite seen on the Slave:

```

08-28 17:54:30.453 [VoIP    ] Prot      <----INVITE sip:Anonymous@14.48.4.88:5060 SIP/2.0
08-28 17:54:30.453 [VoIP    ] Prot      From: "Anonymous" <sip:Anonymous@14.48.4.92:5060;
user=phone>;vnd.pimg.port=1;tag=133B324631353641000BCF02
08-28 17:54:30.453 [VoIP    ] Prot      To: "Anonymous" <sip:Anonymous@14.48.4.88:5060>
08-28 17:54:30.453 [VoIP    ] Prot      Contact: <sip:14.48.4.92:5060>
08-28 17:54:30.453 [VoIP    ] Prot      Content-Type: application/sdp
08-28 17:54:30.453 [VoIP    ] Prot      Supported: replaces, early-session, 100rel
08-28 17:54:30.453 [VoIP    ] Prot      Allow: INVITE, BYE, CANCEL, REFER, NOTIFY, OPTIONS,
REGISTER, INFO, ACK, PRACK
08-28 17:54:30.453 [VoIP    ] Prot      Expires: 120
08-28 17:54:30.453 [VoIP    ] Prot      Call-ID: 02061555D6F5009A000012BC@test.local
08-28 17:54:30.453 [VoIP    ] Prot      CSeq: 1 INVITE

```

```
08-28 17:54:30.453 [VoIP      ] Prot      Max-Forwards:70
08-28 17:54:30.453 [VoIP      ] Prot      User-Agent:PBX-IP Media Gateway
08-28 17:54:30.453 [VoIP      ] Prot      Via:SIP/2.0/UDP 14.48.4.92:5060;
branch=z9hG4bKDC0A05314DD4ED48CEEEA72BD196FC38
08-28 17:54:30.453 [VoIP      ] Prot      Content-Length:245
```

This happens because the calling information is forwarded across the serial cable to the Master TIMG/PIMG, but the Logical Terminal Number (LTN) information does not match up to the port on the T1 Central Authentication Service (CAS) the physical call came in on.

Solution

On TIMG, select **Configuration > Serial > Switch Protocol** in order to configure the Logical Extension Numbers for each port.

Match the TIMG LTN and the port number from the PBX setting. The PBX has a table that shows you which channel on which T1 CAS line uses which LTN. Determine this information from the PBX first and set it accordingly in the TIMG. It is possible to use LTN 1-24 for Master channel 1-24 and LTN 25-48 for Slave channel 1-24.

Related Information

- [TIMG Integration Guide for Cisco Unity Connection Release 9.x](#)
- [PIMG Integration Guide for Cisco Unity Connection Release 9.x](#)
- [TIMG Integration Guide for Cisco Unity Connection Release 10.x](#)
- [PIMG Integration Guide for Cisco Unity Connection Release 10.x](#)
- [Technical Support & Documentation - Cisco Systems](#)