

Unity Connection TIMG/PIMG Calls Roll to the Opening Greeting



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Introduction

T1 Media Gateway (TIMG) and PBX IP Media Gateway (PIMG) devices are used in order to allow PBXs to use Unity Connection as their voicemail system. The communication from TIMG/PIMG to Unity Connection is via Session Initiation Protocol (SIP). The communication from PBX to Unity Connection is via Time-Division Multiplexing (TDM).

This document describes an issue that might be encountered with this type of integration.

Problem

Unity Connection is configured to work with TIMG in order to integrate with PBX. When you call a PBX and Ring No Answer or Call Forward All to voicemail, the call goes to the Opening Greeting instead of to the voicemail greeting.

Traces show:

Note: Some parts of the trace have been re-organized to be easier to view.

TIMG composes a invite because it has received the physical call. However, at this point, TIMG has not received any call information from the PBX.

```
087:57.872 [VoIP      ] Prot    <----INVITE sip:Anonymous@14.48.13.103:5060 SIP/2.0
087:57.872 [VoIP      ] Prot    From: "Anonymous" <sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:57.872 [VoIP      ] Prot    To: "Anonymous" <sip:Anonymous@14.48.13.103:5060>
087:57.872 [VoIP      ] Prot    Contact: <sip:14.48.13.115:5060>
087:57.872 [VoIP      ] Prot    Content-Type: application/sdp
087:57.872 [VoIP      ] Prot    Supported: replaces, early-session, 100rel
087:57.872 [VoIP      ] Prot    Allow: INVITE, BYE, CANCEL, REFER, NOTIFY, OPTIONS,
REGISTER, INFO, ACK, PRACK
087:57.872 [VoIP      ] Prot    Expires: 120
087:57.872 [VoIP      ] Prot    Call-ID: 01B22816147E007E00000019@14.48.13.103
087:57.872 [VoIP      ] Prot    CSeq: 1 INVITE
087:57.872 [VoIP      ] Prot    Max-Forwards: 70
087:57.872 [VoIP      ] Prot    User-Agent: Voice Messaging
087:57.872 [VoIP      ] Prot    Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:57.872 [VoIP      ] Prot    Content-Length: 219
087:57.872 [VoIP      ] Prot
```

```

087:57.872 [VoIP      ] Prot      v=0
087:57.872 [VoIP      ] Prot      o=phone 3397 20425 IN IP4 14.48.13.115
087:57.872 [VoIP      ] Prot      s=-
087:57.872 [VoIP      ] Prot      c=IN IP4 14.48.13.115
087:57.872 [VoIP      ] Prot      t=0 0
087:57.872 [VoIP      ] Prot      m=audio 18698 RTP/AVP 0 101 13
087:57.872 [VoIP      ] Prot      a=rtpmap:0 PCMU/8000/1
087:57.872 [VoIP      ] Prot      a=ptime:30
087:57.872 [VoIP      ] Prot      a=rtpmap:101 telephone-event/8000
087:57.872 [VoIP      ] Prot      a=fmtp:101 0-15
087:57.872 [VoIP      ] Prot      a=rtpmap:13 CN/8000
087:57.872 [VoIP      ] Prot

087:57.872 [VoIP      ] Prot      ---->SIP/2.0 100 Trying
087:57.872 [VoIP      ] Prot      From: "Anonymous" <sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:57.872 [VoIP      ] Prot      To: "Anonymous" <sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
087:57.872 [VoIP      ] Prot      Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:57.872 [VoIP      ] Prot      Expires: 120
087:57.872 [VoIP      ] Prot      Call-ID: 01B22816147E007E00000019@14.48.13.103
087:57.872 [VoIP      ] Prot      CSeq: 1 INVITE
087:57.872 [VoIP      ] Prot      Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
087:57.872 [VoIP      ] Prot      Content-Length: 0
087:57.872 [VoIP      ] Prot
087:57.888 [VoIP      ] Prot

087:57.888 [VoIP      ] Prot      ---->SIP/2.0 180 Ringing
087:57.888 [VoIP      ] Prot      From: "Anonymous" <sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:57.888 [VoIP      ] Prot      To: "Anonymous" <sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
087:57.888 [VoIP      ] Prot      Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:57.888 [VoIP      ] Prot      Expires: 120
087:57.888 [VoIP      ] Prot      Call-ID: 01B22816147E007E00000019@14.48.13.103
087:57.888 [VoIP      ] Prot      CSeq: 1 INVITE
087:57.888 [VoIP      ] Prot      Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
087:57.888 [VoIP      ] Prot      Content-Length: 0
087:57.888 [VoIP      ] Prot
087:57.968 [VoIP      ] Prot

087:57.968 [VoIP      ] Prot      ---->SIP/2.0 200 OK
087:57.968 [VoIP      ] Prot      From: "Anonymous" <sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:57.968 [VoIP      ] Prot      To: "Anonymous" <sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
087:57.968 [VoIP      ] Prot      Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:57.968 [VoIP      ] Prot      Contact: <sip:14.48.13.103:5060>
087:57.968 [VoIP      ] Prot      Expires: 120
087:57.968 [VoIP      ] Prot      Call-ID: 01B22816147E007E00000019@14.48.13.103
087:57.968 [VoIP      ] Prot      CSeq: 1 INVITE
087:57.968 [VoIP      ] Prot      Allow-Events: kpml
087:57.968 [VoIP      ] Prot      Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
087:57.968 [VoIP      ] Prot      Content-Length: 224
087:57.968 [VoIP      ] Prot      Content-Type: application/sdp
087:57.968 [VoIP      ] Prot
087:57.968 [VoIP      ] Prot      v=0
087:57.968 [VoIP      ] Prot      o=CiscoSystemsUCXN 399280213 399280214 IN IP4 14.
48.13.103

```

```

087:57.968 [VoIP      ] Prot    s=No Subject
087:57.968 [VoIP      ] Prot    c=IN IP4 14.48.13.103
087:57.968 [VoIP      ] Prot    t=0 0
087:57.968 [VoIP      ] Prot    m=audio 16716 RTP/AVP 0 101
087:57.968 [VoIP      ] Prot    a=rtpmap:0 PCMU/8000/1
087:57.968 [VoIP      ] Prot    a=ptime:30
087:57.968 [VoIP      ] Prot    a=rtpmap:101 telephone-event/8000
087:57.968 [VoIP      ] Prot    a=fmtp:101 0-15

```

At this point, the call has been answered and the caller hears the Opening Greeting.

```

087:58.448 [VoIP      ] Prot    ---->SIP/2.0 200 OK
087:58.448 [VoIP      ] Prot    From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:58.448 [VoIP      ] Prot    To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
087:58.448 [VoIP      ] Prot    Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:58.448 [VoIP      ] Prot    Contact: <sip:14.48.13.103:5060>
087:58.448 [VoIP      ] Prot    Expires: 120
087:58.448 [VoIP      ] Prot    Call-ID: 01B22816147E007E00000019@14.48.13.103
087:58.448 [VoIP      ] Prot    CSeq: 1 INVITE
087:58.448 [VoIP      ] Prot    Allow-Events: kpml
087:58.448 [VoIP      ] Prot    Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
087:58.448 [VoIP      ] Prot    Content-Length: 224
087:58.448 [VoIP      ] Prot    Content-Type: application/sdp
087:58.448 [VoIP      ] Prot    v=0
087:58.448 [VoIP      ] Prot    o=CiscoSystemsUCXN 399280213 399280214 IN IP4 14.
48.13.103
087:58.448 [VoIP      ] Prot    s=No Subject
087:58.448 [VoIP      ] Prot    c=IN IP4 14.48.13.103
087:58.448 [VoIP      ] Prot    t=0 0
087:58.448 [VoIP      ] Prot    m=audio 16716 RTP/AVP 0 101
087:58.448 [VoIP      ] Prot    a=rtpmap:0 PCMU/8000/1
087:58.448 [VoIP      ] Prot    a=ptime:30
087:58.448 [VoIP      ] Prot    a=rtpmap:101 telephone-event/8000
087:58.448 [VoIP      ] Prot    a=fmtp:101 0-15

```

TIMG has received the call information from PBX. At this point, it is too late since the call has already routed to Unity Connection.

```

087:58.384 [Tel-7     ] Event   Dtmf (#) On
087:58.384 [Tel-7     ] Event   Dtmf (#) Off
087:58.592 [Tel-7     ] Event   Dtmf (0) On
087:58.592 [Tel-7     ] Event   Dtmf (0) Off
087:58.768 [Tel-7     ] Event   Dtmf (2) On
087:58.768 [Tel-7     ] Event   Dtmf (2) Off
087:58.960 [Tel-7     ] Event   Dtmf (#) On
087:58.960 [Tel-7     ] Event   Dtmf (#) Off
087:59.168 [Tel-7     ] Event   Dtmf (5) On
087:59.168 [Tel-7     ] Event   Dtmf (5) Off
087:59.344 [Tel-7     ] Event   Dtmf (2) On
087:59.344 [Tel-7     ] Event   Dtmf (2) Off
087:59.408 [VoIP      ] Prot

087:59.536 [Tel-7     ] Event   Dtmf (8) On
087:59.536 [Tel-7     ] Event   Dtmf (8) Off
087:59.744 [Tel-7     ] Event   Dtmf (6) On
087:59.744 [Tel-7     ] Event   Dtmf (6) Off
087:59.920 [Tel-7     ] Event   Dtmf (#) On
087:59.920 [Tel-7     ] Event   Dtmf (#) Off
088:00.112 [Tel-7     ] Event   Dtmf (5) On
088:00.112 [Tel-7     ] Event   Dtmf (5) Off

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088:00.320 [Tel-7] Event Dtmf (5) On
088:00.320 [Tel-7] Event Dtmf (5) Off
088:00.496 [Tel-7] Event Dtmf (8) On
088:00.496 [Tel-7] Event Dtmf (8) Off
088:00.688 [Tel-7] Event Dtmf (8) On
088:00.688 [Tel-7] Event Dtmf (8) Off
088:00.896 [Tel-7] Event Dtmf (#) On
088:00.896 [Tel-7] Event Dtmf (#) Off
088:01.328 [VoIP] Prot

087:59.408 [VoIP] Prot ---->SIP/2.0 200 OK
087:59.408 [VoIP] Prot From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
087:59.408 [VoIP] Prot To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
087:59.408 [VoIP] Prot Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
087:59.408 [VoIP] Prot Contact: <sip:14.48.13.103:5060>
087:59.408 [VoIP] Prot Expires: 120
087:59.408 [VoIP] Prot Call-ID: 01B22816147E007E00000019@14.48.13.103
087:59.408 [VoIP] Prot CSeq: 1 INVITE
087:59.408 [VoIP] Prot Allow-Events: kpml
087:59.408 [VoIP] Prot Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
087:59.408 [VoIP] Prot Content-Length: 224
087:59.408 [VoIP] Prot Content-Type: application/sdp
087:59.408 [VoIP] Prot
087:59.408 [VoIP] Prot v=0
087:59.408 [VoIP] Prot o=CiscoSystemsUCXN 399280213 399280214 IN IP4 14.48.
13.103
087:59.408 [VoIP] Prot s=No Subject
087:59.408 [VoIP] Prot c=IN IP4 14.48.13.103
087:59.408 [VoIP] Prot t=0 0
087:59.408 [VoIP] Prot m=audio 16716 RTP/AVP 0 101
087:59.408 [VoIP] Prot a=rtpmap:0 PCMU/8000/1
087:59.408 [VoIP] Prot a=ptime:30
087:59.408 [VoIP] Prot a=rtpmap:101 telephone-event/8000
087:59.408 [VoIP] Prot a=fmtp:101 0-15

088:01.328 [VoIP] Prot ---->SIP/2.0 200 OK
088:01.328 [VoIP] Prot From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
088:01.328 [VoIP] Prot To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.328 [VoIP] Prot Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bKC
621E66EBDE4CB89FF423B472071147B
088:01.328 [VoIP] Prot Contact: <sip:14.48.13.103:5060>
088:01.328 [VoIP] Prot Expires: 120
088:01.328 [VoIP] Prot Call-ID: 01B22816147E007E00000019@14.48.13.103
088:01.328 [VoIP] Prot CSeq: 1 INVITE
088:01.328 [VoIP] Prot Allow-Events: kpml
088:01.328 [VoIP] Prot Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
088:01.328 [VoIP] Prot Content-Length: 224
088:01.328 [VoIP] Prot Content-Type: application/sdp
088:01.328 [VoIP] Prot
088:01.328 [VoIP] Prot v=0
088:01.328 [VoIP] Prot o=CiscoSystemsUCXN 399280213 399280214 IN IP4 14.48.
13.103
088:01.328 [VoIP] Prot s=No Subject
088:01.328 [VoIP] Prot c=IN IP4 14.48.13.103
088:01.328 [VoIP] Prot t=0 0
088:01.328 [VoIP] Prot m=audio 16716 RTP/AVP 0 101
088:01.328 [VoIP] Prot a=rtpmap:0 PCMU/8000/1
088:01.328 [VoIP] Prot a=ptime:30
088:01.328 [VoIP] Prot a=rtpmap:101 telephone-event/8000

```
088:01.328 [VoIP      ] Prot      a=fmtp:101 0-15
088:01.920 [Tel-7      ] Event      Tone Detect Enabled (0xFF)
```

After Parsing the Adept rules, the call information from PBX is organized to this statement. This should have been received before the invite so the invite could be composed properly.

```
088:01.920 [Tel-7      ] Event      Cpid (5286,->,->5588,) (NoAns)
088:01.920 [VoIP      ] Prot      <----ACK sip:14.48.13.103:5060 SIP/2.0
088:01.920 [VoIP      ] Prot      CSeq:1 ACK
088:01.920 [VoIP      ] Prot      To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.920 [VoIP      ] Prot      From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
088:01.920 [VoIP      ] Prot      Call-ID:01B22816147E007E00000019@14.48.13.103
088:01.920 [VoIP      ] Prot      Max-Forwards:70
088:01.920 [VoIP      ] Prot      User-Agent:Voice Messaging
088:01.920 [VoIP      ] Prot      Via:SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bK3
032A56F55534D5407B2D30922E6F860
088:01.920 [VoIP      ] Prot      Content-Length:0
088:01.920 [VoIP      ] Prot
088:01.920 [VoIP      ] Prot
```

Here TIMG attempts to act on the updated call information. However, this is not accepted by Unity Connection.

```
088:01.920 [VoIP      ] Prot      <----INFO sip:14.48.13.103:5060 SIP/2.0
088:01.920 [VoIP      ] Prot      Diversion: <tel:5588>;reason=no-answer
088:01.920 [VoIP      ] Prot      Content-Type:text/source-party
088:01.920 [VoIP      ] Prot      To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.920 [VoIP      ] Prot      From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
088:01.920 [VoIP      ] Prot      Call-ID:01B22816147E007E00000019@14.48.13.103
088:01.920 [VoIP      ] Prot      CSeq:2 INFO
088:01.920 [VoIP      ] Prot      Max-Forwards:70
088:01.920 [VoIP      ] Prot      User-Agent:Voice Messaging
088:01.920 [VoIP      ] Prot      Via:SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bK6
EBB9CF7420BA1A393273882D5157413
088:01.920 [VoIP      ] Prot      Content-Length:6
088:01.920 [VoIP      ] Prot
088:01.920 [VoIP      ] Prot      5286
088:01.968 [VoIP      ] Prot

088:01.968 [VoIP      ] Prot      ---->SIP/2.0 200 OK
088:01.968 [VoIP      ] Prot      From: "Anonymous"<sip:Anonymous@14.48.13.115:5060;
user=phone>;vnd.pimg.port=7;tag=42B2324631353641000A6029
088:01.968 [VoIP      ] Prot      To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.968 [VoIP      ] Prot      Via: SIP/2.0/UDP 14.48.13.115:5060;branch=z9hG4bK6
EBB9CF7420BA1A393273882D5157413
088:01.968 [VoIP      ] Prot      Call-ID: 01B22816147E007E00000019@14.48.13.103
088:01.968 [VoIP      ] Prot      CSeq: 2 INFO
088:01.968 [VoIP      ] Prot      Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,
REGISTER,SUBSCRIBE
088:01.968 [VoIP      ] Prot      Content-Length: 0
088:01.968 [VoIP      ] Prot
```

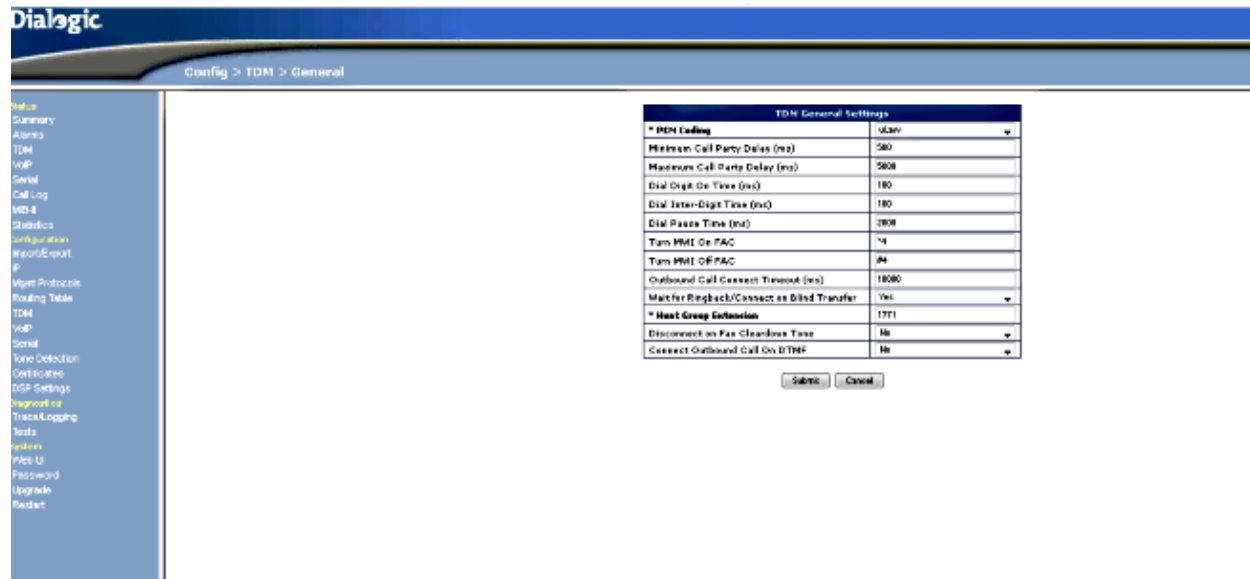
Note: The remainder of the trace has been truncated.

The main issue here is that the call information from the PBX arrives about four seconds after the physical call that triggered the Invite to be sent to Unity Connection is received. Thus the call rolled to the Opening

Greeting because no ***Diversion:*** was contained in the initial invite to Unity Connection to tell Unity Connection to what mailbox to route the call.

Solution

In order to fix this issue, navigate to **Configuration > TDM > General**, find **Maximum Call Party Delay** (ms), and change it to a value such as 5,000 ms. This adds a five-second delay before the initial invite is composed to Unity Connection, which allows time for all call information to be received from PBX.



The screenshot shows the Dialogic configuration interface. The breadcrumb navigation at the top reads "Config > TDM > General". On the left is a vertical navigation menu with various options. The main content area displays a table titled "TDM General Settings".

TDM General Settings	
* REN Calling	Off
Minimum Call Party Delay (ms)	500
Maximum Call Party Delay (ms)	5000
Dial Digit On Time (ms)	100
Dial Inter-Digit Time (ms)	100
Dial Pause Time (ms)	3000
Turn PWT On FAC	No
Turn PWT Off FAC	No
Outbound Call Connect Timeout (ms)	18000
Match Ringback/Connect on Blind Transfer	Yes
* Host Group Extension	1771
Disconnect on Fax Clearhook Tone	No
Connect Outbound Call on DTMF	No

At the bottom of the table are two buttons: "Submit" and "Cancel".