

Rollo de las llamadas del Unity Connection TIMG/PIMG al saludo inicial

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Introducción

El gateway de medios T1 (TIMG) y del gateway de medios IP PBX los dispositivos (PIMG) se utilizan para permitir que los PBX utilicen el Unity Connection como su sistema de correo de voz. La comunicación de TIMG/PIMG al Unity Connection está vía el Session Initiation Protocol (SIP). La comunicación del PBX al Unity Connection está vía el Time-Division Multiplexing (TDM).

Este documento describe un problema que se pudo encontrar con este tipo de integración.

Problema

El Unity Connection se configura para trabajar con TIMG para integrar con el PBX. Cuando usted llama un PBX y Ring No Answer o call forward all al voicemail, la llamada va al saludo inicial en vez al saludo del correo de voz.

Demostración de las trazas:

Nota: Han reorganizado a algunas partes de la traza para ser más fáciles de ver.

TIMG compone una invitación porque ha recibido la llamada física. Sin embargo, en este momento, TIMG no ha recibido ninguna información de la llamada del 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

En este momento, se ha contestado la llamada y el llamador oye el saludo inicial.

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 ha recibido la información de la llamada del PBX. En este momento, es demasiado atrasado puesto que la llamada ha ruteado ya al 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

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)

Después de analizar las reglas del adepto, la información de la llamada del PBX se ordena a esta declaración. Esto debe haber sido recibida antes de que la invitación así que la invitación se podrían componer correctamente.

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

Aquí TIMG intenta actuar en la información de la llamada actualizada. Sin embargo, esto no es validada por el Unity Connection.

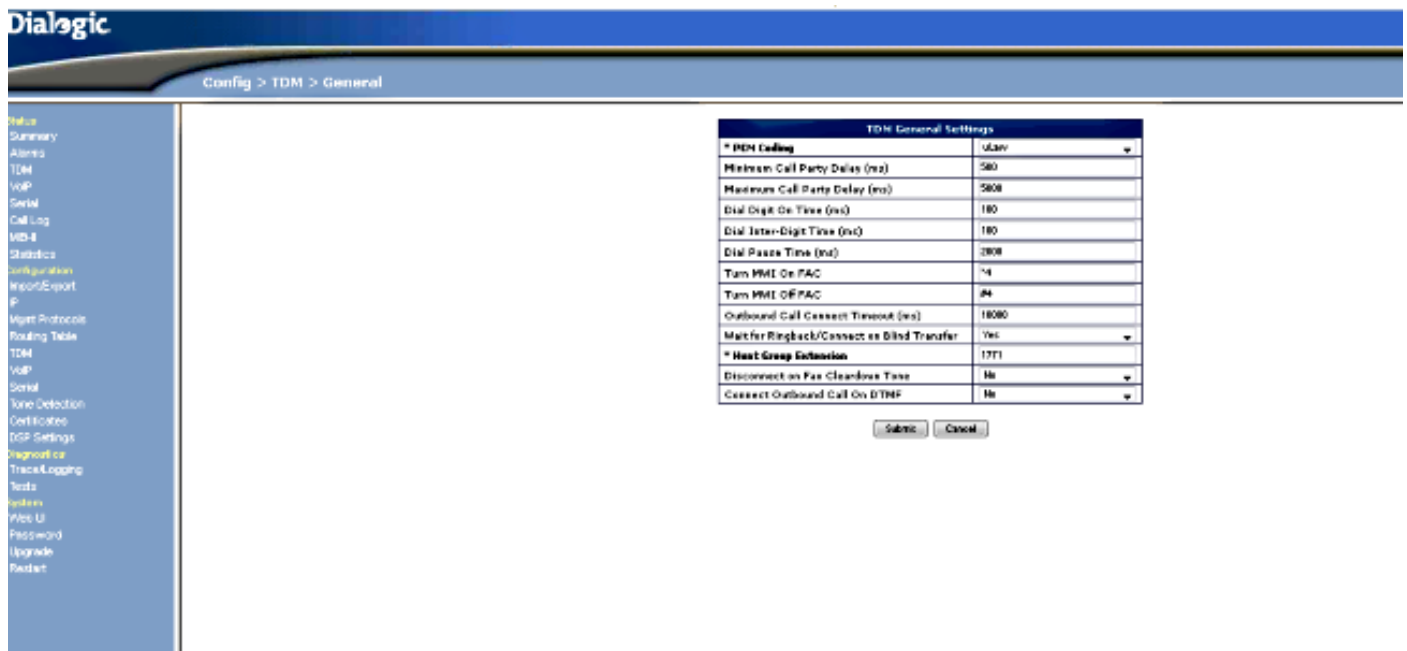
088:01.920 [VoIP] Prot <----INFO sip:14.48.13.103:5060 SIP/2.0
088:01.920 [VoIP] Prot **Diversion: <tel:5588>;reason=no-answer**
088:01.920 [VoIP] Prot Content-Type:text/source-party
088:01.920 [VoIP] Prot To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.920 [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 088:01.920 [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

Nota: El resto de la traza se ha truncado.

La cuestión principal aquí es que llega la información de la llamada del PBX cerca de cuatro segundos después de la llamada física se recibe que accionó la invitación que se enviará al Unity Connection. Así la llamada rodó al saludo inicial porque ninguna **diversión**: fue contenido en la inicial invitan al Unity Connection para decir el Unity Connection a qué buzón para rutear la llamada.

Solución

Para reparar este problema, navegue a la **configuración > al TDM > al general**, encuentran el **retardo máximo del partido de la llamada (ms)**, y lo cambian a un valor tal como ms 5,000. Esto agrega un cinco-segundo retardo antes de que la inicial invite se componga al Unity Connection, que permite que la hora para que toda la información de la llamada sea recibida del PBX.



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

TDM General Settings	
* IPM Calling	uJan
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)	2000
Turn PWI On FAC	Y
Turn PWI Off FAC	N
Outbound Call Connect Timeout (ms)	18000
Wait for Ringback/Connect on Blind Transfer	Yes
* Host Group Extension	1771
Disconnect on Fax Cleanline Tone	No
Connect Outbound Call On DTNF	No

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