Unity Connection TIMG/PIMG コールがオープニング グリーティングを始める

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概要

PBX が自身のボイスメール システムとして Unity Connection を使用できるように、T1 メディアゲートウェイ（TING）デバイスと PBX IP メディアゲートウェイ（PING）デバイスが使用されます。 TING/PING から Unity Connection への通信は、Session Initiation Protocol（SIP）経由で行われます。 PBX から Unity Connection への通信は、時分割多重（TDM）経由で行われます。

このドキュメントでは、このタイプの統合で発生する可能性のある問題について説明します。

問題

PBX と統合するために TING で動作するように Unity Connection が設定されています。 PBX に発信し、無応答またはボイスメールへの不在転送になった場合、その通話がボイスメールの挨拶ではなく、オープニング グリーティングに転送されます。

トレースには次のように表示されます。

> **注:** トレースの一部は、閲覧しやすいように再編成されています。

実際に通話を受け取ったために TING が招待を作成します。 ただし、この時点では、TIMG は PBX からいずれの通話情報も受信していません。

```
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
```

この時点で、通話に応答があり、発信者にはオープニング グリーティングが流れます。

```
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
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 が PBX から通話情報を受信しました。 この時点では、通話がすでに Unity Connection にルーティングされているので遅すぎます。

```
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)
```

Adept ルールを解析後、PBX からの通話情報が、このステートメントに編成されます。 招待が適切に構成されるように、このステートメントを招待の前に受け取る必要があります。

```
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
```

ここで TING は、更新された通話情報を実行しようとします。 ただし、Unity Connection はこれを受け付けません。

```
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
```

注: トレースの残りの部分は切り捨てられています。

ここでの主な問題は、Unity Connection に送信される招待をトリガーする実際の通話を受けてから約 4 秒後に、PBX からの通話情報を受け取る点です。 Unity Connection にどのメールボックスに通話をルーティングするのかを指示する Unity Connection への最初の招待に、**Diversion:** が 含まれていないために、通話がオープニング グリーティングに転送されてしまいます。

解決策

この問題を解決するには、[Configuration] > [TDM] > [General] に移動し、[Maximum Call Party Delay (ms)] を探して、5,000 ms などの値に変更します。 これにより、Unity Connection に対して最初の招待が作成される前に 5 秒の遅延が追加されるので、PBX からすべての通話情報を受け取る時間的余裕が生まれます。