

Rolo dos atendimentos da conexão de unidade TIMG/PIMG à saudação inicial

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Introdução

O gateway de mídia T1 (TIMG) e do gateway de mídia IP PBX dispositivos (PIMG) são usados a fim permitir que os PBX usem a conexão de unidade como seu sistema de correio de voz. A comunicação de TIMG/PIMG à conexão de unidade é através do Session Initiation Protocol (SIP). A comunicação do PBX à conexão de unidade é através do Time-Division Multiplexing (TDM).

Este documento descreve uma edição que possa ser encontrada com este tipo de integração.

Problema

A conexão de unidade é configurada para trabalhar com TIMG a fim integrar com PBX. Quando você chama um PBX e Ring No Answer ou call forward all ao correio de voz, o atendimento vai à saudação inicial em vez à saudação de correio de voz.

Mostra dos traços:

Nota: Algumas partes do traço foram reorganizadas para ser mais fáceis de ver.

TIMG compõe um convite porque recebeu o atendimento físico. Contudo, neste momento, TIMG não recebeu nenhuma informação de chamada do 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
```

Neste momento, o atendimento foi respondido e o chamador ouve a saudação 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 recebeu a informação de chamada do PBX. Neste momento, está demasiado atrasado desde que o atendimento tem distribuído já à conexão de unidade.

```
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)

Após ter analisado gramaticalmente as regras do adepto, a informação de chamada do PBX é organizada a esta indicação. Isto deveu ter sido recebido antes que o convite assim que o convite poderiam ser compostos corretamente.

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

Aqui TIMG tenta atuar na informação de chamada actualizado. Contudo, isto não é aceiteado pela conexão de unidade.

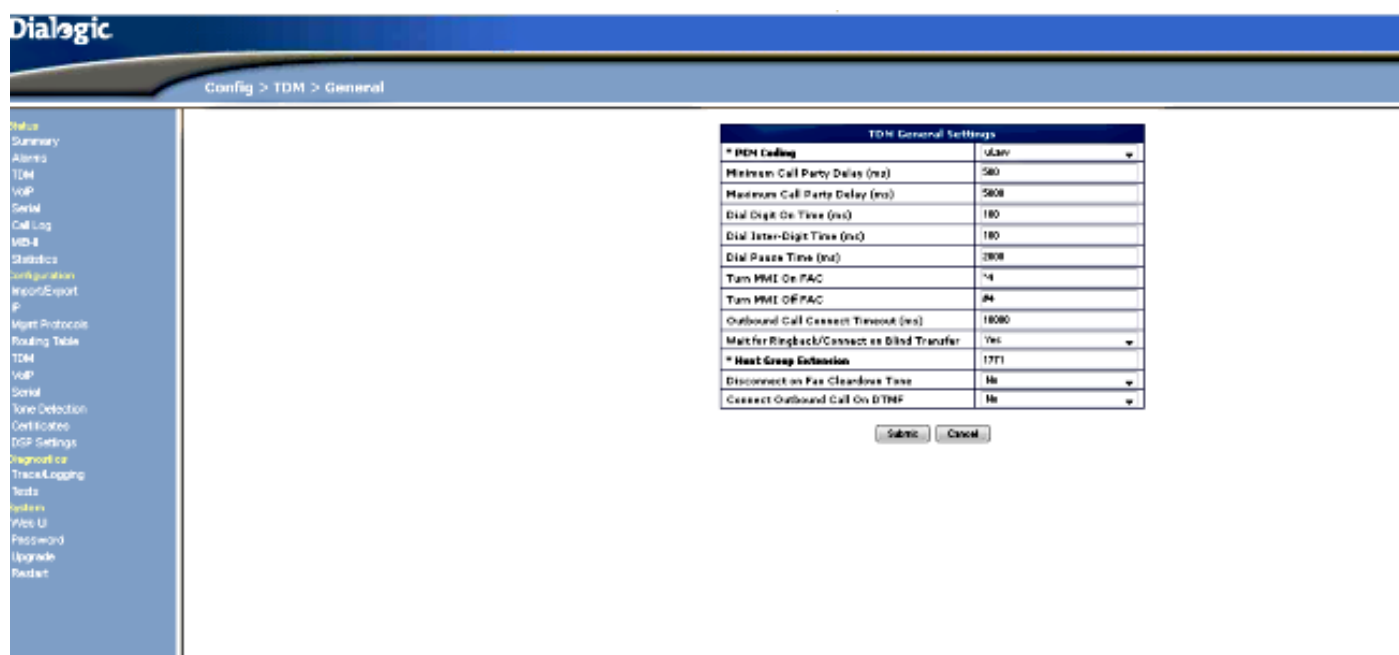
088:01.920 [VoIP] Prot <----INFO sip:14.48.13.103:5060 SIP/2.0
088:01.920 [VoIP] Prot **Diversion: <tel:5588>;reason=no-answer**
088:01.920 [VoIP] Prot Content-Type:text/source-party
088:01.920 [VoIP] Prot To: "Anonymous"<sip:Anonymous@14.48.13.103:5060>;
tag=f0c09771bd2942e7a57794619f8efccd
088:01.920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

A questão principal aqui é que a informação de chamada do PBX chega aproximadamente quatro segundos após o atendimento físico que provocou o convite a ser enviado à conexão de unidade é recebido. Assim o atendimento rolou à saudação inicial porque nenhuma **diversão**: foi contido na inicial convidam à conexão de unidade para dizer a conexão de unidade a que caixa postal para distribuir o atendimento.

Solução

A fim fixar esta edição, navegar à **configuração > ao TDM > ao general**, encontrar o **atraso máximo do partido do atendimento** (Senhora), e mudá-lo a um valor tal como a Senhora 5,000. Isto adiciona um cinco-segundo atraso antes que a inicial convide for composta à conexão de unidade, que permite que a hora para que toda a informação de chamada esteja recebida do 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 | |
|---|----------|
| * PRM Calling | U.S.A. ▼ |
| 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 PWE On/FAC | Y |
| Turn PWE Off/FAC | N |
| Outbound Call Connect Timeout (ms) | 10000 |
| Make for Ringback/Connect as Blind Transfer | Yes ▼ |
| * Next Group Extension | 1771 |
| Disconnect on Fax Cleardown Tone | No ▼ |
| Connect Outbound Call on DTMF | No ▼ |

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