Dialogic® 1000 and 2000 Media Gateway Series

SIP Compliance (Version 6.0 SU8 Software)

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Revision History

Revision	Release date	Notes
05-2666-004	September 2011	Version 6.0 SU8 Software
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Refer to www.dialogic.com for product updates and for information about support policies, warranty information, and service offerings.

1. Scope

This document describes the implementation and usage of the Session Initiated Protocol (SIP) by the Dialogic® 1000 Media Gateway Series (DMG1000) and Dialogic® 2000 Media Gateway Series (DMG2000) product lines, which also are referred to collectively herein as Dialogic® Media Gateway or Media Gateway or gateway. A gateway can provide a connection between VOIP telephony networks and proprietary digital telephony networks.

This document assumes that the reader understands the function and operation of the gateway (see the Dialogic® 1000 and 2000 Media Gateway Series User's Guide) and is familiar with SIP.

2. References

Dialogic® 1000 and 2000 Media Gateway Series User's Guide

3. Configuration Parameters

The following table lists the configuration parameters that govern the operation of the SIP mode of the gateway. This is not a complete list of all gateway configurable parameters.

See the Dialogic® 1000 and 2000 Media Gateway Series User's Guide for a list that was complete as of its date of publication.

Configuration INI Parameters

The following SIP configurations are stored in text format in the gateway file named config.ini.

Parameter (INI)	Valid Settings	Default	Description
dspDigitRelay	None, RFC2833, Inband-Tone	RFC2833	RTP Digit Relay Mode.
dspFaxModemToneRelay	RFC2833, Inband-Tone	RFC2833	RTP Fax/Modem Tone Relay Mode.
dspLBRCodec	G.723.1, G.729AB	G.723.1	Selects the Low Bit Rate Codec.
dspVAD	On, Off	On	Voice Activity Detection.
gwFaxTransportMode	T.38, G.711 Passthrough, None	T.38	Defines fax-transport method.
gwIsdnIe77Enable	Yes, No	No	When enabled (set to 'Yes'), the Gateway will propagate a received ISDN information element 77h to a SIP INVITE message. The data is sent as a section of a multipart MIME message. The information element is propagated only when it is received in an ISDN SETUP message.
gwMonitorCallConns	Yes, No	No	When enabled (set to 'Yes'), the Media Gateway will monitor the connection

Parameter (INI)	Valid Settings	Default	Description
			state of active IP calls. If the active IP call has lost connection, the Media Gateway will tear down the call.
gwMonitorCallIntSec	10-3600	30	Call Monitor Interval (secs).
gwMonitorVoipHostsIntSec	10-3600	30	Host Monitor Interval (secs).
gwProactiveMonitorDnsARecordEnable	Yes, No	No	When the Proactive DNS Monitoring is enabled (set to 'Yes'), the Gateway will monitor the IP addresses obtained from DNS resolution. If an IP address does not respond to SIP OPTIONS pings, the gateway does not attempt to send it SIP INVITES. When the IP address resumes responding to SIP OPTIONS pings, the gateway resumes sending it SIP INVITES.
gwQosCallControl	0-255	0	Call Control QoS Byte.
gwQosRtp	0-255	0	RTP QoS Byte.
gwRTPEndPort	6000-65000	50000	RTP Start Port.
gwRTPStartPort	6000-65000	49000	RTP Start Port.
gwRTPValidateSrcIp	On, Off	Off	RTP Source IP Address Validation
gwRTPValidateSrcPort	On, Off	Off	RTP Source UDP Port Validation.
gwSigDigitRelay	On, Off	Off	Signaling Digit Relay Mode.

Parameter (INI)	Valid Settings	Default	Description
gwU2UEnable	On, Off	Off	Enables/disables gateway support for the user-user header. See chapter 15.
gwU2UTranslateMethod	String, Hex	String	If the user-user header does not specify an encoding, this parameter is used as the implied encoding.
secSipTlsProtocol	SSLv3_TLSv1, SSLv3_Only, TLSv1_Only	SSLv3_TLSv1	SSL TLS Protocol.
secSipTlsUseSelfSignedCert	Self-Signed, CA-Signed	Self-Signed	TLS Certificate Type.
sipAcceptableMedia	RTP_Only, SRTP_Only, RTP_SRTP	RTP_SRTP	Acceptable Media on INVITE reception.
sipCallAsDomainName	No, Yes	No	Defines the host name used in the From header of generated INVITEs. If 'Yes', the gateway's SIP Server Domain is used as the host name. If 'No', the gateway's VOIP address is used as the host name.
sipDnsServerAddr	IPv4 dotted decimal address.	Blank	IP Address in dotted decimal notation.
sipDnsServerAddr2	IPv4 dotted decimal address.	Blank	IP Address in dotted decimal notation (secondary).
sipEarlyMediaSupport	Always, On-Demand, None	OnDemand	RFC 3960 Early Media Support.
sipEnumDnsEnabled	Yes, No	No	DNS Translation of Phone Numbers.

Parameter (INI)	**Valid Settings**	**Default**	**Description**
sipExpInvSec	0-60000	120	Invite Expiration Time (secs).
sipPrivacyHdrEnabled	Yes, No	No	Enable SIP Privacy headers (RFC3325 & RFC3323, and draft-ietf-sip-privacy-04
sipPrivacyHdrMethod	P-Asserted-Identity Remote-Party-ID Both	P-Asserted-Identity	Supported headers
sipRelProvRsp	None, Required, Supported	Supported	Reliable Provisional Responses.
sipRetryAfter	1–60000	60	SIP Retry After hint (sec).
sipSendSupportedSDPMedia	Yes, No	No	Send Supported SDP Media Types.
sipServerDomain	String with length between 1-254 characters.	pbxgw.default.com	Host and domain name.
sipServerPort	1024-65000	5060	TCP/UDP Server Port.
sipSipsUriEnabled	Yes, No	No	SIPS URI Scheme.
sipT1Multiplier	1-255	64	T1 Multiplier.
sipT1TimeMs	100-60000	500	T1 timer (msecs) – Request Retransmit Start Time.
sipT2TimeMs	200-60000 (must be greater than T1)	4000	T2 timer (msecs) – Request Retransmit Max Time.
sipT4TimeMs	1000-60000	5000	T4 timer. (msecs) – Specifies amount of time the network will take to clear messages between client and server transactions.

Parameter (INI)	Valid Settings	Default	Description
sipTcpInactivitySec	10-60000	90	Number of seconds after which an idle TCP connection will be closed.
sipTlsCertVerifyDate	Yes, No	Yes	Verify TLS Peer Certificate Date.
sipTlsCertVerifyPurpose	Yes, No	Yes	Verify TLS Peer Certificate Purpose.
sipTlsCertVerifyTrust	Yes, No	Yes	Verify TLS Peer Certificate Trust.
sipTlsCipherListType	RSA, RSA_NULL_ENCRYPTION	RSA	Type of cipher list used by SIP TLS.
sipTlsEnabled	Yes, No	No	TLS Transport Enabled.
sipTlsInactivitySec	10-60000	30	TLS Inactivity Timer (sec).
sipTlsMutualAuthentication	Yes, No	Yes	Mutual TLS Authentication Required.
sipTlsServerPort	1024-65000	5061	TLS Server Port.
sipUdpTcpEnabled	Yes, No	Yes	UDP/TCP Transports Enabled.

Configuration XML parameters

The following SIP configuration items are stored in XML format in the gateway file named dmg.xml.

Parameter (XML)	**Valid Settings**	**Default**	**Description**
RouteTable/ VoipHostGroups/ Group/**Host**	Valid endpoint address (VOIP, URL, alias, or blank)	Blank	Specifies a VOIP endpoint that is to receive calls from the either the TDM or VOIP network.
RouteTable/ VoipHostGroups/ Group/**FaultTolerant**	Yes, No	No	Enables/Disables fault-tolerant handling of outbound VOIP calls. If 'Yes', then the Media Gateway will failover to the next configured VOIP endpoint if an outbound VOIP call attempt fails. If 'No', then a failed outbound VOIP call attempt will not be retried.
RouteTable/ VoipHostGroups/ Group/**LoadBalance**	Yes, No	No	Enables/Disables load-balancing of outbound VOIP calls. If 'Yes', then the Media Gateway will route outbound VOIP calls to each configured VOIP endpoint in a round-robin fashion. If 'No', then the Media Gateway will not load-balance across multiple VOIP endpoints.
SIP/ NetworkGroups/ Group/ Signaling/ **Transport**	TCP, UDP, TLS	UDP	Preferred Call Signal transport to be used on all initial requests sent by the gateway.
SIP/	Yes, No	No	Applicable only if 'Transport Type' is configured for TLS.

Parameter (XML)	Valid Settings	Default	Description
NetworkGroups/ Group/ Signaling/ **SIPS**			Yes = All Request, To, From, and Contact URIs generated by the gateway will use the SIPS URI scheme. No = All Request, To, From, and Contact URIs generated by the gateway will use the SIP URI scheme.
SIP/NetworkGroups/Group/Signaling/**UserPhone**	Yes, No	Yes	Specifies if the user=phone parameter is specified in the SIP URI for TDM calls.
SIP/NetworkGroups/Group/Signaling/ **PhoneContextLocal**	String with length between 1–128 characters.	Blank	Specifies the phone-context of the SIP URI on the From: line. If blank, then no phone-context is specified.
SIP/NetworkGroups/Group/Signaling/ **PhoneContextRemote**	String with length between 1–128 characters.	Blank	Specifies the phone-context of the SIP URI on the To: line. If blank, then no phone-context is specified.
SIP/NetworkGroups/Group/Proxy/Server/**Addr**	String with length between 1–128 characters.	Blank	The SIP URI of the Primary Proxy Server through which the Gateway may send/receive requests. If blank, the Gateway will not use the Primary Proxy Server.
SIP/NetworkGroups/Group/Proxy/Server/**Port**	1024-65000	5060	The IP Port of the SIP Proxy Server.
SIP/NetworkGroups/Group/Proxy/**Query**	10-36000	30	Interval at which the Proxy Server(s) are queried. The Proxy Server must respond to a SIP OPTIONS request in order for the proxy query to succeed. If the Primary Proxy Server does not respond to the query, the gateway switches to the Backup Proxy Server. Once the Primary Proxy Server responds to the query, the gateway switches back to the Primary Proxy Server.
SIP/NetworkGroups/Grou	String with length	Blank	IP Address of the SIP Registration Server that the Gateway should

Parameter (XML)	**Valid Settings**	**Default**	**Description**
p/Register/**Addr**	between 1–128 characters.		register with. If blank, the Gateway will not register with a Registration Server.
SIP/NetworkGroups/Group/Register/**Port**	1024-65000	5060	IP Port of the SIP Registration Server.
SIP/NetworkGroups/Group/Register/**User**	String with length between 1-64 characters.	Blank	Specifies the User-Field of the address-of-record to be registered. If blank, then no User-Field is specified.
SIP/NetworkGroups/Group/Register/**GwName**	String with length between 1-64 characters.	Blank	Specifies the Gateway Name in the SIP Register message. If blank the gateway's IP address is used.
SIP/NetworkGroups/Group/Register/**Expire**	10-60000	120	Specifies the value to be placed in the Expires header of transmitted INVITE requests. If the value is zero, an Expires header is not added to the INVITE request.
SIP/NetworkGroups/Group/Audio/**Codec**	G.711u, G.711a, G.723.1, G.729AB	G.711u, G.711a	Audio codec preference selection.
SIP/NetworkGroups/Group/Audio/**G711PacketSize**	10,20, 30	30	Size of a G.711 codec audio packet. Packet size = (Frame Size * Frames Per Packet).
SIP/NetworkGroups/Group/Audio/**G723PacketSize**	30, 60	30	Size of a G.723 codec audio packet. Packet size = (Frame Size * Frames Per Packet)
SIP/NetworkGroups/Group/Audio/**G729PacketSize**	10,20,30,40,50,60	30	Size of a G.729 codec audio packet. Packet size = (Frame Size * Frames Per Packet).
SIP/NetworkGroups/Group/SRTP/**Preference**	None, RTP_Only, SRTP_Only	RTP_Only	Determines how SRTP is negotiated for a media session.

Parameter (XML)	**Valid Settings**	**Default**	**Description**
SIP/NetworkGroups/Group/SRTP/**AuthTag**	32, 80	80	Length of the SHA1 authentication tag in bits.
SIP/NetworkGroups/Group/SRTP/**TxMKI**	Yes, No	Yes	If enabled then a one-byte MKI (Master Key Index) is used when transmitting secure-RTP and secure-RTPC packets. If disabled, MKI is not used.
SIP/NetworkGroups/Group/SRTP/**KDREnable**	Yes, No	Yes	If enabled then a new encryption key is derived after 2^KDR packets. If disabled, a key is derived only at the start of the transmission.
SIP/NetworkGroups/Group/SRTP/**KDR**	16-24	16	Key derivation rate for outbound INVITE'S. This value is the exponent of a power of 2.
SIP/NetworkGroups/Group/SRTP/**WSH**	64-99	64	Anti-replay window size hint.
SIP/NetworkGroups/Group/SRTP/**UnEncryptSRTP**	Yes, No	No	If enabled, then transmitted RTP VOICE packets will NOT be encrypted despite the negotiation of cipher keys.
SIP/NetworkGroups/Group/SRTP/**UnEncryptSRTCP**	Yes, No	No	If enabled, then transmitted RTCP CONTROL packets will NOT be encrypted despite the negotiation of cipher keys.
SIP/NetworkGroups/Group/SRTP/**UnAuthSRTP**	Yes, No	No	If enabled, then RTP VOICE packets will NOT be authenticated.
SIPAuth/Authentication /**Enabled**	Yes, No	No	Enables Authentication for Inbound SIP Calls.
SIPAuth/Authentication /**GwRealm**	String with length between 1-64 characters.	default.gw.com	Sets the Gateway Realm for Inbound SIP calls.
SIPAuth/Authentication/ Users/User/**UserName**	String with length between 1-64 characters.	Blank	Specifies an acceptable User Name associated with **GwRealm** for inbound SIP methods.

Parameter (XML)	**Valid Settings**	**Default**	**Description**
SIPAuth/Authentication/ Users/User/**Password**	String with length between 1-64 characters.	Blank	Specifies the Password associated with **UserName** for inbound SIP methods.
SIPAuth/Authentication /**Algorithm**	MD5, MD5-sess	MD5	Sets the Authentication algorithm.
SIPAuth/Authentication/ Methods/Method/**Name**	Invite, Register, Notify, Info, Bye, Refer, Options	N/A	Identifies the SIP method which may be Authenticated. There is one entry for each SIP method (valid setting).
SIPAuth/Authentication/ Methods/Method/ **Challenge**	Yes, No	No	Determines if the specified SIP method (**Name**) will be Authenticated on receive. There is one entry for each SIP method (valid setting).
SIPAuth/Authorization /**Enabled**	Yes, No	No	Enables Authorization for outbound SIP methods.
SIPAuth/Authorization /Realms**/RealmName**	String with length between 1-64 characters.	Blank	Defines the Realm to be associated with outbound SIP methods.
SIPAuth/Authorization /Realms**/UserName**	String with length between 1-64 characters.	Blank	Defines the User Name to be associated with **RealmName** for outbound SIP methods.
SIPAuth/Authorization /Realms**/Password**	String with length between 1-64 characters.	Blank	Defines the Password to be associated with **UserName** for outbound SIP methods.

4. General

VOIP to TDM Connect Determination

The gateway does not send the 200 OK response to the INVITE until a positive detect event is detected on the TDM network call. The positive detect event could be voice activity, ring-back tone termination, etc. In this case, the gateway waits for the TDM network call to be answered before the 200 OK response is sent and the bi-directional media channel is established. If the gateway detects error or busy tone on the TDM network call during call origination, then the gateway terminates the TDM network call and responds with a failure to the INVITE request. This is generally preferred if the gateway is to be used by a voice-mail system or other 'automated' system that requires positive answer verification on the TDM network.

SDP Usage

The gateway supports G.711, G.723.1 and G.729AB codecs. The codec preference is set in the gateway's configuration.

The SDP information placed in INVITE requests reflects the configured codec preference. The RTP channels used by the gateway are also configurable. When SRTP is supported, SDP INVITE also contains the media type, either RTP/SAVP for secured or RTP/AVP for non-secured. For RTP/SAVP media type, crypto information is provided including encryption algorithm, authentication type and tag, master Key Index (MKI), key derivation rate (KDR), and window size hint (WSH).

On INVITE requests generated by the gateway, the gateway sends the desired RTP transmit port as well as the codec preferences. The gateway expects the 200 OK response from the far party to contain the far party's transmit RTP port as well as its codec preference. The gateway will use the first codec specified in the far party's preference list that matches a codec in the gateway's codec preference list. If no match is found, then the gateway disconnects the call. This should not happen since the far party should not send a 200 OK if there are no matching codec preferences.

On INVITE requests received by the gateway, the gateway expects the SDP information to include the requesting party's codec preferences. If none of the preferences match the gateway's capabilities, or the INVITE request does not specify an RTP port, then the gateway responds with a 488 Not Acceptable error. The gateway will use the first codec specified in the far party's preference list that matches a codec in the gateway's codec preference list. If no SDP is included in the INVITE, the gateway may respond with a 200 OK which contains the gateway's SDP offer.

A VOIP hold request is performed by the gateway by using the a=sendonly attribute. The gateway will accept a hold request from a peer through either the a=sendonly attribute or via setting `c=IN IP4 0.0.0.0` in the SDP of a re-INVITE request (to which the gateway will respond with an a=sendonly attribute). The gateway stops all RTP transmit for the call. The un-hold request is handled by setting `c=IN IP4 VOIP-gateway` in the SDP of a re-INVITE request or by removing the a=sendonly attribute. The gateway continues RTP transmit for the call.

5. Functions

Function	Supported?
User Agent Client (UAC)	Yes
User Agent Server (UAS)	Yes
Proxy Server	3rd Party
Redirect Server	Yes

6. Supported RFCs and Drafts

Specification	Notes
RFC 3261 SIP	None
RFC 3263 DNS Resolution	None
RFC 2976 SIP INFO Method	None
RFC 3824 Using ENUM for SIP Applications	None
RFC 4028: The SIP Session Timer	None
RFC 3265: SIP – Specific Event Notification	None
RFC 3515: The REFER Method	Used only for call transfer.
Draft-ietf-sipping-cc-transfer-07: Call Transfer	None
RFC 3892: The Referred-By Mechanism	None
RFC 3891: The SIP Replaces Header	None
RFC 3264: An Offer Answer Model with Session Description	None
ITU-T Recommendation T.38: Procedures for real-time Group 3 facsimile communication over IP networks	Only UDPTL transport is supported.
RFC 3842: A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol	None
RFC 3326: The Reason Header for the Session Initiation Protocol	None
RFC 3959: The Early Session Disposition Type for the Session Initiation Protocol	Application model support. No support for Gateway model.
RFC 3960: Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)	Application model support. No support for Gateway model.
RFC 3262: Reliability of Provisional Responses in Session Initiation Protocol (SIP)	Supports reliable provisional responses and reception of PRACK, but does require them.
RFC 4568: Security Descriptions for Media Streams	None
RFC 3398: ISDN ISUP to SIP Mapping	None
RFC 4566: Session Description Protocol	None
RFC 2617: HTTP Authentication	Used for SIP Authentication. Basic Authentication is not supported for SIP. Digest

	Authentication is supported
Draft-johnston-cuss-sip-uui-01.pdf	Using the transport mechanism described in section 3.6

7. Methods

Method	Supported?	Comments
INVITE	Yes	Generated and received for call initiation and hold/unhold.
ACK	Yes	Generated and received.
OPTIONS	Yes	Generated to monitor status of proxy server and VOIP endpoints (when **RouteTable/VoipHostGroups/Group/FaultTolerant** is enabled).
BYE	Yes	Generated and received.
INFO	Yes	Generated and received. Generated on TDM to VOIP call party update.
CANCEL	Yes	Generated and received.
REGISTER	Yes	Generated for gateway registration and ignored if received.
REFER	Yes	Generated and received for call transfer.
NOTIFY	Yes	Generated and received for transfer and MWI.
SUBSCRIBE	Yes	Received, but ignored.
MESSAGE	Yes	Received, but ignored.
PRACK	Yes	Generated if a 183 response is received with 100rel.

8. Responses

1xx Response – Information Responses

Response	Supported?	Comments
100 Trying	Yes	Generated for an incoming INVITE.
180 Ringing	Yes	Generated when ring-back tone is detected on outbound TDM call.
181 Call is being forwarded	Yes	Not generated. Ignored on receive.
182 Queued	Yes	Not generated. Ignored on receive.
183 Session Progress	Yes	Generated and received. Generated in response to early media request or call progress from TDM side.

2xx Response – Successful Responses

Response	Supported?	Comments
200 OK	Yes	Generated and received
202 Accepted	Yes	Generated and received (REFER, NOTIFY).

3xx Response – Redirection Responses

Response	Supported?	Comments
300 Multiple Choices	Yes	Not generated. If received, Contact list is traversed.
301 Moved Permanently	Yes	Not generated. If received, Contact list is traversed.
302 Moved Temporarily	Yes	Generated if inbound SIP call is to be redirected to another IP address due to dial plan rules. If received, Contact list is traversed.
305 Use Proxy	Yes	Not generated. If received, Contact list is traversed.
380 Alternate Service	Yes	Not generated. If received, call gracefully fails/disconnects.

4xx Response – Request Failure Responses

Response	Supported?	Comments
400 Bad Request	Yes	Generated on receive of invalid requests. When received, call gracefully fails/disconnects.
401 Unauthorized	Yes	Generated when UAS Authentication is enabled. When received, re-sends the method with the Authorization header if UAC Authentication is enabled.
402 Payment Required	Yes	Not generated. When received, call gracefully fails/disconnects.
403 Forbidden	Yes	Generated on receive of INVITE that cannot be routed to a telephony port. When received, call gracefully fails/disconnects.
404 Not Found	Yes	Generated when Special Information Tone (SIT) is detected for reasons of "operator intercept" or "vacant circuit" TDM. Also generated if an MWI request to the TDM network fails because of an unknown extension number. When received, call gracefully fails/disconnects.
405 Method Not Allowed	Yes	Generated on receive of a non-supported method. When received, call gracefully fails/disconnects.
406 Not Acceptable	Yes	Generated on receive of any method which fails SIP authentication. When received, call gracefully fails/disconnects.
407 Proxy Authentication Required	Yes	Not generated. When received, re-sends the method with the Authorization header if UAC Authentication is enabled.
408 Request Timeout	Yes	Not generated. When received, call gracefully fails/disconnects.
409 Conflict	Yes	Not generated. When received, call gracefully fails/disconnects.
410 Gone	Yes	Not generated. When received, call gracefully fails/disconnects.
411 Length Required	Yes	Not generated. When received, call gracefully fails/disconnects.
413 Request Entity Too Large	Yes	Not generated. When received, call gracefully fails/disconnects.
414 Request URL Too Long	Yes	Generated. When received, call gracefully

Response	Supported?	Comments
		fails/disconnects.
415 Unsupported Media	Yes	Not generated. When received, call gracefully fails/disconnects.
420 Bad Extension	Yes	Not generated. When received, call gracefully fails/disconnects.
480 Temporarily Unavailable	Yes	Generated when call can be routed to TDM network, but the telephony port is already active. Generated when outbound PSTN call or PSTN MWI set/clear fails because of glare condition. When received, call gracefully fails/disconnects.
481 Call Leg/Transaction/Subscription Does Not Exist	Yes	Generated and received. When received, call gracefully fails/disconnects.
482 Loop Detected	Yes	Not generated. When received, call gracefully fails/disconnects.
483 Too Many Hops	Yes	Not generated. When received, call gracefully fails/disconnects.
484 Address Incomplete	Yes	Not generated. When received, call gracefully fails/disconnects.
485 Ambiguous	Yes	Not generated. When received, call gracefully fails/disconnects.
486 Busy Here	Yes	Generated when busy tone detected on TDM. When received, call gracefully fails/disconnects.
487 Request Canceled	Yes	Generated upon receiving a CANCEL for a pending INVITE. When received, call gracefully disconnects.
488 Not Acceptable	Yes	Generated if a received INVITE contains no supported media types. When received, call gracefully fails/disconnects.
489 Bad Event	Yes	Generated if a NOTIFY is received for an event that the gateway does not support. When received, the corresponding NOTIFY request fails.

5xx Response – Server Failure Responses

Response	Supported?	Comments
500 Internal Server Error	Yes	Generated if internal Server error occurs. When received, call gracefully fails/disconnects.

501 Not Implemented	Yes	Not generated. When received, call gracefully fails/disconnects.
502 Bad Gateway	Yes	Not generated. When received, call gracefully fails/disconnects.
503 Service Unavailable	Yes	Generated on detection of Special Information Tone (SIT) for reasons of "reorder" or "no circuit found" on TDM. When received, call gracefully fails/disconnects.
504 Gateway Timeout	Yes	Not generated. When received, call gracefully fails/disconnects.
505 Version Not Supported	Yes	Not generated. When received, call gracefully fails/disconnects.
513 Message Too Large	Yes	Not generated. When received, call gracefully fails/disconnects.

6xx Response – Global Responses

Response	Supported?	Comments
600 Busy Everywhere	Yes	Not generated. When received, call gracefully fails/disconnects.
603 Decline	Yes	Not generated. When received, call gracefully fails/disconnects.
604 Does Not Exist Anywhere	Yes	Not generated. When received, call gracefully fails/disconnects.
606 Not Acceptable	Yes	Not generated. When received, call gracefully fails/disconnects.

9. Headers

All supported headers in the table are supported in the receive direction. Those generated as well are noted.

Header	Supported?	Comments
Accept	Yes	
Accept-Encoding	Yes	
Accept-Language	Yes	
Allow	Yes	
Allow-Events	Yes	
Authentication-Info	No	
Also	No	
Authorization	Yes	Generated
Call-ID	Yes	Generated
Contact	Yes	Generated
Content-Disposition	No	
Content-Encoding	Yes	Generated
Content-Language	No	
Content-Length	Yes	Generated
Content-Type	Yes	Generated
Cseq	Yes	Generated
Date	Yes	
Diversion	Yes	Generated
Encryption	No	
Error-Info	No	
Event	Yes	
Expires	Yes	Generated
From	Yes	Generated
In-Reply-To	No	
Join	Yes	Generated
Max-Forwards	Yes	
MIME-Version	No	Generated
Min-Expires	Yes	
Organization	Yes	
P-Asserted-Identity	Yes	Generated

Priority	Yes	
Privacy	Yes	Generated
Proxy-Authenticate	Yes	
Proxy-Authorization	Yes	
Proxy-Require	Yes	
Reason	Yes	Generated
Record-Route	Yes	
Remote-Party-ID	Yes	Generated
Reply-To	No	

Referred-By	Yes	Generated
Referred-To	Yes	Generated
Replaces	Yes	Generated
Requested-By	Yes	Generated
Require	Yes	
Response-Key	No	
Retry-After	Yes	
Route	Yes	Generated
Server	Yes	Generated
Subject	Yes	
Subscription-State	Yes	
Supported	Yes	
Timestamp	Yes	
To	Yes	Generated
Unsupported	Yes	
User-Agent	Yes	Generated
User-To-User	Yes	Generated
Via	Yes	Generated
Warning	Yes	
WWW-Authenticate	Yes	Generated

10. SIP METHODS

INVITE Requests (Generated)

From Header

The *From* header of the INVITE request identifies the TDM network originator of the call. INVITE requests originating from the gateway provide source party information, including the phone number of the TDM network calling party (if available) and the telephony port (logical TDM channel) on which the TDM network call was received. For instance, if TDM network extension 211 originated the call to the gateway on telephony port 2 (logical TDM channel 2), the *From* header of the INVITE would appear as follows (assumes that the VOIP address of the gateway is set to 10.10.11.1):

```
From: “211” <sip:211@10.10.11.1>;vnd.pimg.port=2
```

If the TDM network originating party contains both number and name information, the number will be placed in the user field and the name will be placed in the display field, as follows:

```
From: “Joe Smith” <sip:211@10.10.11.1>;vnd.pimg.port=2
```

If the TDM network originating party cannot be determined, then the originating party is set to ‘Anonymous’, as follows:

```
From: “Anonymous” <sip:Anonymous@10.10.11.1>;vnd.pimg.port=2
```

If Media Gateway OEM code is set to 1002, then the port information will be placed in the user field, as follows:

```
From: “Anonymous” <sip:port2@10.10.11.1>
```

Contact Header

The sip address of the *Contact* header of the INVITE request contains the VOIP address and VOIP port of the gateway to which all future requests should be sent. If ***sipServerPort*** is configured as 5060, then the *Contact* header appears as (assuming the gateway VOIP address is 10.10.11.1):

```
Contact: sip:10.10.11.1:5060
```

If the **SIP/NetworkGroups/Group/Signaling/Transport** is set to TCP, then the transport is added as a uri-parameter:

```
Contact: <sip:10.10.11.1:5060;Transport=TCP>
```

To Header

The *To* header is created using the configured **RouteTable/VoipHostGroups/Group/Host.**

If the **RouteTable/VoipHostGroups/Group/Host** is an IP address, then the user name 'Anonymous' is used. For instance, if **RouteTable/VoipHostGroups/Group/Host** is set to '10.10.1.132', then the *To* header appears as:

```
        To: "Anonymous" <sip:Anonymous@10.10.1.132>
```

If **RouteTable/VoipHostGroups/Group/Host** is a URI, such as 'sipinet.com', then the user name 'Anonymous' is used:

```
        To: <sip:Anonymous@sipnet.com>
```

If the **RouteTable/VoipHostGroups/Group/Host** is a URI, such as 'joe@10.10.1.132', then the entire **RouteTable/VoipHostGroups/Group/Host** is used:

```
        To: <sip:joe@10.10.1.132>
```

If the **RouteTable/VoipHostGroups/Group/Host** is not a URI and not an IP address, then a proxy is required. In this case, **RouteTable/VoipHostGroups/Group/Host** is assumed to be a user name, and its value is pre-pended to the proxy address to create a URI. For instance, if **RouteTable/VoipHostGroups/Group/Host** is set to '200', and the proxy address is 10.10.1.100, then the *To* header appears as:

```
        To: <sip:200@10.10.1.100>
```

If the called-party number received from the inbound TDM call is '211', and **RouteTable/VoipHostGroups/Group/Host** is 'voicemail.sipnet.com', then the To header will look as follows:

```
To: <sip:211@voicemail.sipnet.com>
```

If the called-party number received from the inbound TDM call is '211', and **RouteTable/VoipHostGroups/Group/Host** is a URL with a user-name, then the called-party is not used. For instance, if **RouteTable/VoipHostGroups/Group/Host** is 'joe@voicemail.sipnet.com', then the To header will look as follows:

```
        To: <sip:joe@voicemail.sipnet.com>
```

Diversion Header

The *Diversion* header is used to convey the original TDM network destination party of a TDM network call that has been diverted to the gateway. This may occur if the original TDM network destination is busy, doesn't answer, or is in a forwarded condition. This header is present if the TDM call has been diverted (forwarded). If the TDM call is direct, then the *Diversion* header is not present. If the *Diversion* header is present, it will contain the original TDM network destination extension, in the form of a tel URL. The header also contains the reason for the diversion.

For instance, if the original TDM network destination party '211', diverted to the gateway because of a busy condition, the *Diversion* header appears as follows:

```
        Diversion: <tel:211> ;reason="user-busy"
```

Supported Diversion reasons consist of the following:

“no-answer”, “unconditional”, “user-busy”

Route Header

The generated INVITE request will contain a Route header if **SIP/NetworkGroups/Group/Proxy/Server/Addr** is set. The Route header will contain the IP address of the currently-active proxy server specified by **SIP/NetworkGroups/Group/Proxy/Server/Addr.** This will cause the INVITE request to be sent to the proxy-server before being sent to the destination specified in the To header.

For instance, the following INVITE request is being sent to 102 at necdsi.swdev.lab, but it is first sent to the proxy-server at IP address 172.16.6.14, as specified in the Route header.

```
INVITE sip:102@necdsi.swdev.lab SIP/2.0
From:"2008"<sip:2008@172.16.4.6:5060>;vnd.pimg.port=1;tag=0FB6324631353641000196A
To:sip:102@necdsi.swdev.lab
Content-Type:application/sdp
Supported:replaces,timer
Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,REFER,SUBSCRIBE,NOTIFY,
MESSAGE
Session-Expires:30
Min-SE:10
Call-ID:01B2270F1B81400000000006@ntm1_dni.swdev.lab
CSeq:1 INVITE
Route:<sip:172.16.6.14:5060;lr>
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Contact:sip:2008@172.16.4.6:5060
Via:SIP/2.0/UDP 172.16.4.6:5060;branch=z9hG4bK1ED51D2E1D29C72955C0E4F753F199C5
Content-Length:296
v=0
o=phone 17383 0 IN IP4 172.16.4.6
s=-
c=IN IP4 172.16.4.6
t=0 0
m=audio 49012 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
m=image 0 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPRedundancy
```

INVITE Requests (Received)

Vendor-Specific Port Specifier

When a VOIP call is received by the gateway, the gateway routes the call based on the gateway's configured Routing Table. If a TDM port is specified in the INVITE sent to the gateway, using the vendor specific 'vnd.pimg.port' header-parameter, then the Gateway will attempt to use the specified TDM port to originate a TDM call. Use of the 'vnd.pimg.port' header-parameter is not recommended as its use requires the application to track the availability of Gateway TDM channels.

An INVITE may explicitly specify the desired gateway telephony port (logical TDM channel) to which the call is to be routed. This telephony port may be specified as a vendor specific attribute in the *To* header.

Although not recommended, the application can specify the desired TDM channel by adding the vnd.pimg.port header parameter to the *To* header.

```
To: <sip:211@10.10.11.1>;vnd.pimg.port=3
```

Determination of Number to Dial

The gateway requires a TDM network phone number to dial on the TDM network. The number to dial is extracted from the received INVITE request message.

The gateway examines the INVITE request message for the dialable number in the following order:

1) *Request-URI* user (tel or sip) – if it is a dialable number

2) *To* header user (tel or sip) – if it is a dialable number

3) *To* header display name – if it is a dialable number

4) Finally, the dialable number may be modified or even generated based on Routing Table rules.

Early-Media

VOIP-to-TDM Calls

A SIP endpoint that wants to receive RTP on an outbound connection to the TDM network from the gateway prior to the TDM subscriber answering the call may request early-media. If the gateway's **sipEarlyMediaSupport** is configured for On-Demand, then the INVITE that is received by the gateway must have a Supported header that contains early-session. The INVITE must also contain SDP with an active and compatible audio stream. The gateway will then send a 183 response to the INVITE and will send RTP to the connection address prior to the 200OK response to the INVITE. The following is an example of an INVITE that is sent to the gateway requesting early-media.

```
INVITE sip:305@172.16.4.3:5060 SIP/2.0
Via: SIP/2.0/UDP 172.16.2.25:5061;branch=z9hG4bK-1-0
From: <sip:172.16.2.25:5061>;tag=1
To: <sip:305@172.16.4.3:5060>
Call-ID: 1-2012@172.16.2.25
CSeq: 1 INVITE
Contact: sip:172.16.2.25:5060
Max-Forwards: 70
Subject: Early Media Test
Content-Type: application/sdp
Supported: early-session
Content-Length:  134
v=0
o=user1 53655765 2353687637 IN IP4 172.16.2.25
s=-
c=IN IP4 172.16.2.25
t=0 0
m=audio 10000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

If the gateway's **sipEarlyMediaSupport** is configured for Always then early media is enabled for all VOIP to TDM calls (regardless of the presence or absence of 'early-session' from the client) provided that the INVITE contains SDP with an active and compatible audio stream. The gateway will then send a 183 response to the INVITE and will send RTP to the connection address prior to the 200OK response to the INVITE.

TDM-to-VOIP Calls

If the gateway’s **sipEarlyMediaSupport** is configured for On-Demand, then the INVITE that is sent by the gateway will include the Supported header that includes ‘early-session’. The INVITE will also contain SDP with an active and compatible audio stream. If the VOIP peer sends a 183 in response to the INVITE which includes the SDP answer, the gateway will send RTP to the connection address prior to the 200OK response. The following is an example of an INVITE that is sent by the gateway requesting early-media.

```
INVITE sip:105@172.16.3.32;transport=tcp SIP/2.0
From:<sip:103@172.16.3.108:5060;user=phone>;vnd.pimg.port=1;tag=3DC832463135364100000F6A
To:<sip:105@172.16.3.32>
Content-Type:application/sdp
Supported: early-session,100rel
Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,PRACK,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Expires:120
Call-ID:01B2270E1C81400000000000@pbxgw.default.com
CSeq:1 INVITE
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Contact:sip:103@172.16.3.108:5060;transport=tcp
Via:SIP/2.0/TCP 172.16.3.108:5060;branch=z9hG4bK960F02532790A8DD24EB62D2C279135C
Content-Length:208
v=0
o=phone 7485 0 IN IP4 172.16.3.108
s=-
c=IN IP4 172.16.3.108
t=0 0
m=audio 49000 RTP/AVP 0 8 101 13
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

If the gateway’s **sipEarlyMediaSupport** is configured for Always, then the INVITE that is sent by the gateway will contain SDP with an active and compatible audio stream. If the VOIP peer sends a 183 response which includes the SDP answer, the gateway will send RTP to the connection address prior to the 200OK.

Positive Answering Machine Detection (PAMD)

A SIP endpoint that sends an INVITE to the gateway may request the gateway to perform a DSP analysis of the voice of the answering party. This is known as a PAMD request. The purpose of the PAMD request is to allow the peer to know if the source of the audio is human, answering machine or unknown. A new SDP attribute is defined for reporting PAMD.

The attribute is **X-audio-source** and is a value attribute (as defined in RFC 4566). The possible PAMD result values are the following:

- **answering-machine**,
- **human**
- **unknown**

The attribute is **X-pamd-on** and is a binary attribute (as defined in RFC 4566). This is a media-level attribute. As a result of the INVITE below, the gateway checks for PAMD on the call.

[Offer]

```
v=0
o=phone 22945 0 IN IP4 172.16.3.74
s=
c=IN IP4 172.16.3.74
t=0 0
m=audio 49012 RTP/AVP 0 8 101 13
a=rtpmap:8 PCMA/8000/1
a=fmtp:101 0-15
a=X-pamd-on
```

When the Gateway responds to the INVITE with a 200OK, it will echo the X-pamd-on request in the SDP. An ACK is expected as normal from the peer. The answer to the PAMD request is not known at this time. It will take the DSP of the Gateway several seconds to determine the source of the voice. When the analysis is complete, the Gateway will send a re-INVITE to the peer. The SDP content will be identical to that which was contained in the 200OK with the addition of the PAMD result specified in the X-audio-source.

The following is an example of an SDP message reporting that an answering machine was detected on the audio stream.

```
v=0
o=phone 22945 0 IN IP4 172.16.3.74
s=
c=IN IP4 172.16.3.74
t=0 0
m=audio 49012 RTP/AVP 0 8 101 13
a=rtpmap:8 PCMA/8000/1
a=fmtp:101 0-15
a=X-pamd-on
a=X-audio-source:answering-machine
```

INVITE Responses

The gateway will send a 200OK response to a received INVITE request only upon receiving a connect event from the switch (here we are considering only VOIP and TDM routing and not VOIP to VOIP routing). A connect event signals that the destination party of the outbound TDM call has answered the call from the gateway. A connect event can be signaled by the presence of voice on the TDM channel, the breaking of ring-back tone, an ISDN CONNECT message, or a connect-timeout.

While the gateway is waiting for a connect event, the outbound call to the TDM switch may be rejected by the switch. The call-rejection event from the switch is translated into a SIP error response that is sent as a response to the INVITE. A call-rejection event may be a tone (such as busy tone, error, congestion) or an ISDN DISCONNECT message. The following table contains the mappings from a TDM-switch call-rejection event and the INVITE response generated by the gateway. Note that the mapping from ISDN DISCONNECT message 'cause' codes to SIP responses follows RFC3398 recommendations.

Rejection Event	***SIP Response Generated***
Busy Tone	486 Busy Here
Congestion Tone	404 Not Found
Error Tone	404 Not Found
DISCONNECT, Cause Code = 1 unallocated number	404 Not Found
DISCONNECT, Cause Code = 2 no route to network	404 Not Found
DISCONNECT, Cause Code = 3 no route to destination	404 Not Found
DISCONNECT, Cause Code = 17 user busy	486 Busy Here
DISCONNECT, Cause Code = 18 no user responding	408 Request Timeout
DISCONNECT, Cause Code = 19 no answer from the user	480 Temporarily unavailable
DISCONNECT, Cause Code = 20 subscriber absent	480 Temporarily unavailable
DISCONNECT, Cause Code = 21 call rejected	403 Forbidden
DISCONNECT, Cause Code = 22 number changed	410 Gone
DISCONNECT, Cause Code = 23 redirection to new destination	410 Gone
DISCONNECT, Cause Code = 26 non-selected user clearing	404 Not Found
DISCONNECT, Cause Code = 27 destination out of order	502 Bad Gateway
DISCONNECT, Cause Code = 28 address incomplete	502 Bad Gateway
DISCONNECT, Cause Code = 29 facility rejected	501 Not implemented
DISCONNECT, Cause Code = 31 normal unspecified	480 Temporarily unavailable
DISCONNECT, Cause Code = 34 no circuit available	503 Service unavailable

DISCONNECT, Cause Code = 38 network out of order	503 Service unavailable
DISCONNECT, Cause Code = 41 temporary failure	503 Service unavailable
DISCONNECT, Cause Code = 42 switching equipment congestion	503 Service unavailable
DISCONNECT, Cause Code = 47 resource unavailable	503 Service unavailable
DISCONNECT, Cause Code = 55 incoming calls barred within CUG	403 Forbidden
DISCONNECT, Cause Code = 57 bearer capability not authorized	403 Forbidden
DISCONNECT, Cause Code = 58 bearer capability not presently available	503 Service unavailable
DISCONNECT, Cause Code = 65 bearer capability not implemented	488 Not Acceptable Here
DISCONNECT, Cause Code = 70 only restricted digital avail	488 Not Acceptable Here
DISCONNECT, Cause Code = 79 service or option not implemented	501 Not implemented
DISCONNECT, Cause Code = 87 user not member of CUG	403 Forbidden
DISCONNECT, Cause Code = 88 incompatible destination	503 Service unavailable
DISCONNECT, Cause Code = 102 recovery of timer expiry	504 Gateway timeout

INFO

The INFO method is generated by the gateway if the TDM call party information was not available when the INVITE request was made, but now is available. This could occur if the call party information did not arrive at the gateway at the same time as the TDM call, or if the TDM call had to be answered before its call party information is made available by the TDM switch.

In this case, the calling party information is placed in the body of the message (*Content-Type* header is set to 'text/source-party'), and the *Diversion* header (if call has been diverted) is populated as described in section 0.

If TDM call party information was not available when the INVITE request was made, and is found to be never available, then the gateway may (if the gateway parameter ***gwInformIfNoPbxCpid*** is true) generate an INFO message to the VOIP endpoint with no new call information (no diversion header or source party information in the body of the message) when the ***gwInformIfNoPbxCpidMs*** timer expires.

The following shows an INFO request sent with 3211 as the new calling-party number:

```
INFO sip:102@necdsi.swdev.lab SIP/2.0
From:<sip:Anonymous@172.16.4.6:5060>;vnd.pimg.port=1;tag=0FB6324631353
To:sip:102@necdsi.swdev.lab;tag=0FBDKD837323A
Content-Type:text/source-party
Supported:replaces,timer
Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Session-Expires:30
Min-SE:10
Call-ID:01B2270F1B81400000000006@ntm1_dni.swdev.lab
CSeq:1 INVITE
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Contact:sip:2008@172.16.4.6:5060
Via:SIP/2.0/UDP 172.16.4.6:5060;branch=z9hG4bK1ED51D2E1D29C72
Content-Length:4

3211
```

The Media Gateway is able to process/generate SIP INFO message for DTMF tone relay. SIP INFO message is sent along the signaling path of the call. Upon receipt of a SIP INFO message, with DTMF relay content, the gateway will generate specified DTMF tone on the TDM trunk line. The opposite will be true as well. If DTMF relay signaling is enabled in the gateway(INI parameter: **gwSigDigitRelay**), a SIP INFO message will be generated with DTMF relay content every time a digit is pressed on the TDM side of the call.

The SIP INFO method is used by a UA to send call signaling information to another UA with which it has an established media session. The following example shows a SIP INFO message with DTMF content:

```
INFO sip:7007471000@example.com SIP/2.0
Via: SIP/2.0/UDP alice.uk.example.com:5060
From: <sip:7007471234@alice.uk.example.com>;tag=d3f423d
To: <sip:7007471000@example.com>;tag=8942
Call-ID: 312352@myphone
CSeq: 5 INFO
Content-Length: 24
Content-Type: application/dtmf-relay
Signal=5
Duration=160
```

This sample message shows a SIP INFO message received by the gateway with specifics about the DTMF tone to be generated. The combination of the From, To, and Call-ID headers identifies the call leg. The signal and duration headers specify the digit, in this case 5, and duration, 160 milliseconds.

Media Gateway already has a configuration parameter for the duration of a DTMF digit (INI parameter: **telDialDigitOnMs**). If this value is set then it will be used as default. Otherwise the “Duration” header value will be used. If duration is longer than 5000 ms, the maximum duration of 5000 ms is used as default. If duration is less than 100 ms, the minimum duration of 100 ms is used as default. If, for some reason, duration is not specified, default signal duration will be set to 250 ms.

BYE

The BYE method is generated to terminate a call with a UA on the VOIP network. The Reason header contains the reason for the call termination if it can be determined. For instance, if dial-tone was detected on the line, the Reason header would appear as:

```
Reason:      E.182;text=”Dialtone detected ; duration = 1100ms"
```

The duration field is present if the disconnect event is a tone. This provides the recipient of the BYE request the amount of time the gateway took to detect the disconnect event.

The following table contains the reason text that may be reflected in the Reason header.

Text	***Duration Provided?***	***Description***
"Dialtone Detected"	Yes	Dial-tone was detected during a connected call.
"Disconnect Tone Detected"	Yes	Disconnect tone was detected.
"Congestion Tone Detected"	Yes	Congestion tone was detected.
"Busy Tone Detected"	Yes	Busy tone was detected.
"Error Tone Detected"	Yes	Error tone was detected.
"Loop Current Disconnect"	No	Loop current disconnect was detected.
"Dtmf Disconnect Detected"	Yes	A DTMF disconnect string was detected.
"Special Info Tone Detected"	Yes	Special-info tone was detected.
"Normal"	Yes	Out of band disconnect message received from switch.

CANCEL

The CANCEL method is generated to cancel a call from the TDM network and the VOIP network.

OPTIONS

The OPTIONS Request is generated when the Gateway is monitoring the status of a proxy-server or the status of a VOIP endpoint (**RouteTable/VoipHostGroups/Group/FaultTolerant** is enabled). The following OPTIONS request is an example of a request that is sent to the primary proxy server or a VOIP endpoint.

```
OPTIONS sip:Anonymous@172.16.6.14:5060 SIP/2.0
To:sip:Anonymous@172.16.6.14:5060
From:sip:172.16.4.6;tag=00CA32463135364100001B96
Contact:sip:172.16.4.6
Supported:replaces
Call-ID:01B2270F5381400000000007@ntm1_dni.swdev.lab
CSeq:1 OPTIONS
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Via:SIP/2.0/UDP 172.16.4.6:5060;branch=z9hG4bK5C5CD655AD08F6C4893F2AA6CF9229D2
Content-Length:0
```

Proxy-Server Monitoring

When a primary and a backup outbound proxy server is configured

(**SIP/NetworkGroups/Group/Proxy/Server/Addr)** and an outbound call via the proxy server fails, the Gateway will send periodic OPTIONS requests to the primary proxy to monitor its health. The call-flow is described in section 12. . The primary proxy server must respond to the OPTIONS request (either with a 200 OK or an error-response) in order for the Gateway to switch back to using the primary proxy server. An error-response is sufficient since not all proxy-servers support the reception of an OPTIONS request. It is assumed that the proxy-server, when it is healthy, will at least respond to the OPTIONS request with an error, if not a 200 OK.

VoIP Endpoint Monitoring

VoIP endpoint monitoring is enabled when:

1. Fault Tolerant is enabled on the Routing Table's VoIP Host Groups page OR
2. The configuration parameter 'gwProactiveMonitorDnsARecordsEnable' is enabled AND when the endpoint was obtained from the DNS resolution of an FQDN.

For the remainder of this section, the phrase 'when VoIP Endpoint Monitoring is enabled' means that one of the preceding conditions is true.

When VoIP EndPoint Monitoring is enabled the Gateway will send OPTIONS requests to the VOIP server endpoint (peer) on a periodic basis until the peer responds with a '200 OK' or a "503 Service Unavailable'. It is expected that the peer, at a minimum, replies to an OPTIONS request. Therefore the Gateway requires that the peer responds with a '200 OK' or a '503 Service Unavailable' before the Gateway will again send it any calls.

Fault-Tolerant Failover

Failover will take place whenever the peer is incapable of responding to an SIP OPTIONS message. Therefore, if the Gateway receives a 4xx or 5xx (except 503) error message from the peer in response to OPTIONS, failover will take place.

When operating in TCP mode, the Gateway must attempt to open and connect a TCP socket with the
peer in order to send the OPTIONS. The failure to connect the TCP socket does not directly trigger failover however the failover result will be the same since the peer was incapable of responding to the aborted OPTIONS request.

Failover may also occur even when the peer is responding to the OPTIONS. If the peer is actively responding to the OPTIONS but does not respond to an INVITE after roughly T1(default = 500 ms)) * T1 Multi(default = 64) seconds then the call will failover to the next host which will then be sent the INVITE.

Additionally, if the peer is responding OK to the OPTIONS but sends an error response to the INVITE (such as a 4xx or 5xx) then the INVITE will be sent to the next host in the list.

NOTIFY (“message-summary”)

If the gateway receives a NOTIFY request with the *Event* header to “message-summary”, the gateway assumes this to be a request to set/clear a TDM network MWI. The *To* header of the indication must contain the dial number of the TDM network destination of the MWI set/clear (see section 0) The *Messages-Waiting* header must be set to either “yes” (if the MWI is to be set on) or “no” (if the MWI is to be set off). The media-type is ignored.

The Gateway will not SUBSCRIBE to “message-summary” events. The Gateway supports the reception of un-subscribed NOTIFY requests. NOTIFY requests sent to the Gateway must contain a Subscription-State value of “terminated”. Since the “message-summary” NOTIFY may create an implicit subscription within the gateway, the NOTIFY request MUST contain a Subscription-State header (as defined by RFC 3265) with a value of “terminated” so that an implicit subscription is not created within the gateway.

The following is a NOTIFY request received by the gateway to set the message-waiting indicator of subscriber 2222 to an ‘ON’ state.

```
NOTIFY sip:2222@172.16.3.56:5060 SIP/2.0
From:<sip:172.16.3.137:5060>;tag=5816324631353641006B5CB0
To:"2222"<sip:2222@172.16.3.56:5060>
Event:message-summary
Content-Type:application/simple-message-summary
Call-ID:01B231C907814000000000C3@gw.default.com
CSeq:1 NOTIFY
Allow-Events:refer,message-summary
Via:SIP/2.0/UDP 172.16.3.137:5060
Contact:<sip:172.16.3.137:5060>
User-Agent:Test Application
Max-Forwards:70
Supported:100rel,timer,replaces
Subscription-State:terminated
Content-Length:23
Messages-Waiting: yes
```

REFER

The REFER method indicates that the recipient (identified by Request-URI) should contact a third party using the contact information provided in the request. Media Gateway uses REFER method for doing supervised and un-supervised transfers. If the "Refer-To" of the REFER request contains the IP address of the current gateway, then the REFER is requesting a transfer of a call that is on current gateway to another call that is on the same gateway. The transfer is in this case handled via a circuit transfer, not an IP transfer. This assumes that the URI in the Refer-To header contains an IP address and not a DNS name. If a DNS name is used, then this check will fail. The Refer-To header may contain embedded headers, such as "Replaces", that must be passed onto the transfer destination when the INVITE is generated to the destination. The "Replaces" embedded header contains the call-id of the call to replace. If the Refer-To points to the gateway, then the "Replaces" embedded header is checked to see if it specifies another call on the current gateway. If it does, then this is a transfer complete request for a supervised transfer. If replaces header is not present then unsupervised transfer is done. Below is a REFER message request that the gateway will process to perform an unsupervised (blind) transfer.

```
REFER sip:8128569221@129.79.50.17:5060 SIP/2.0
To: sip:8128569221@129.79.50.17:5060>;vnd.pimg.port=22;tag=10C53246313536410000036E
From: <sip:8128560786@129.79.50.18>;tag=26599
Call-ID: 01B2270CE181400000000004@voip-gw02.uits.indiana.edu
CSeq:2 REFER
Via: SIP/2.0/UDP 129.79.50.18;branch=z9hG4bK4c7c694f950f2aefaaad51976
Contact: <sip:8128560786@129.79.50.18>
Refer-To: <sip:61202@129.79.50.17:5060>
Referred-By: <sip:8128560786@129.79.50.18>
Max-Forwards: 70
Supported: join, replaces
User-Agent: ININ/2.400.10.12657
Content-Length: 0
```

REGISTER

The REGISTER method may be configured to be sent for each Network Group specified. On reception, the gateway may respond with a 200OK (if authentication passes or is disabled) however the gateway does not store or pass on received registrations.

The REGISTER message is formed as follows using abbreviations based on user interface data. Only the relevant fields are shown:

REGISTER	sip:**RSA**:**RSP**
From:	sip:**RU**@**GWN**
To:	sip:**RU**@**GWN**
Contact:	<sip:**RU**@**CIP**:**CSP**>
Expires:	**EXP**

The abbreviations are identified in the table below.

Abbreviation	User Interface Name	Parameter (from XML or INI)
RSA	Registration Server Address	SIP/NetworkGroups/Group/Register/**Addr**
RSP	Registration Server Port	SIP/NetworkGroups/Group/Register/**Port**
RU	Registered User	SIP/NetworkGroups/Group/Register/**User**
GWN	Gateway Name	SIP/NetworkGroups/Group/Register/**GwName**
EXP	Registration Expiration	SIP/NetworkGroups/Group/Register/**Expire**
CIP	Client IP Address	**ipClientAddr**
CSP	TCP/UDP Server Port	**sipServerPort**

The qualifiers regarding the use of these variables is as follows:

1. The Registration Server Address can be any string, including a Fully Qualified Domain Name (FQDN) or IP address.
2. The Registered User can be any string.
3. The Gateway Name can be any string but if it is blank then the gateway's (Client) IP address is used.
4. If the Registration Server Address is empty then registration is not attempted.
5. The gateway will perform only one REGISTER per Network Group. It will not attempt multiple registrations based upon TDM channels or extensions.

6. Some Registrars may require the Gateway Name be set to the Registration Server Address.

REGISTER	From	To	Contact
RSA:RSP	RU@GWN	RU@GWN	RU@CIP:CSP
	GWN (when RU is null)	GWN (when RU is null)	CIP:CSP (when RU is null)

An example REGISTER sent from the gateway is shown here.

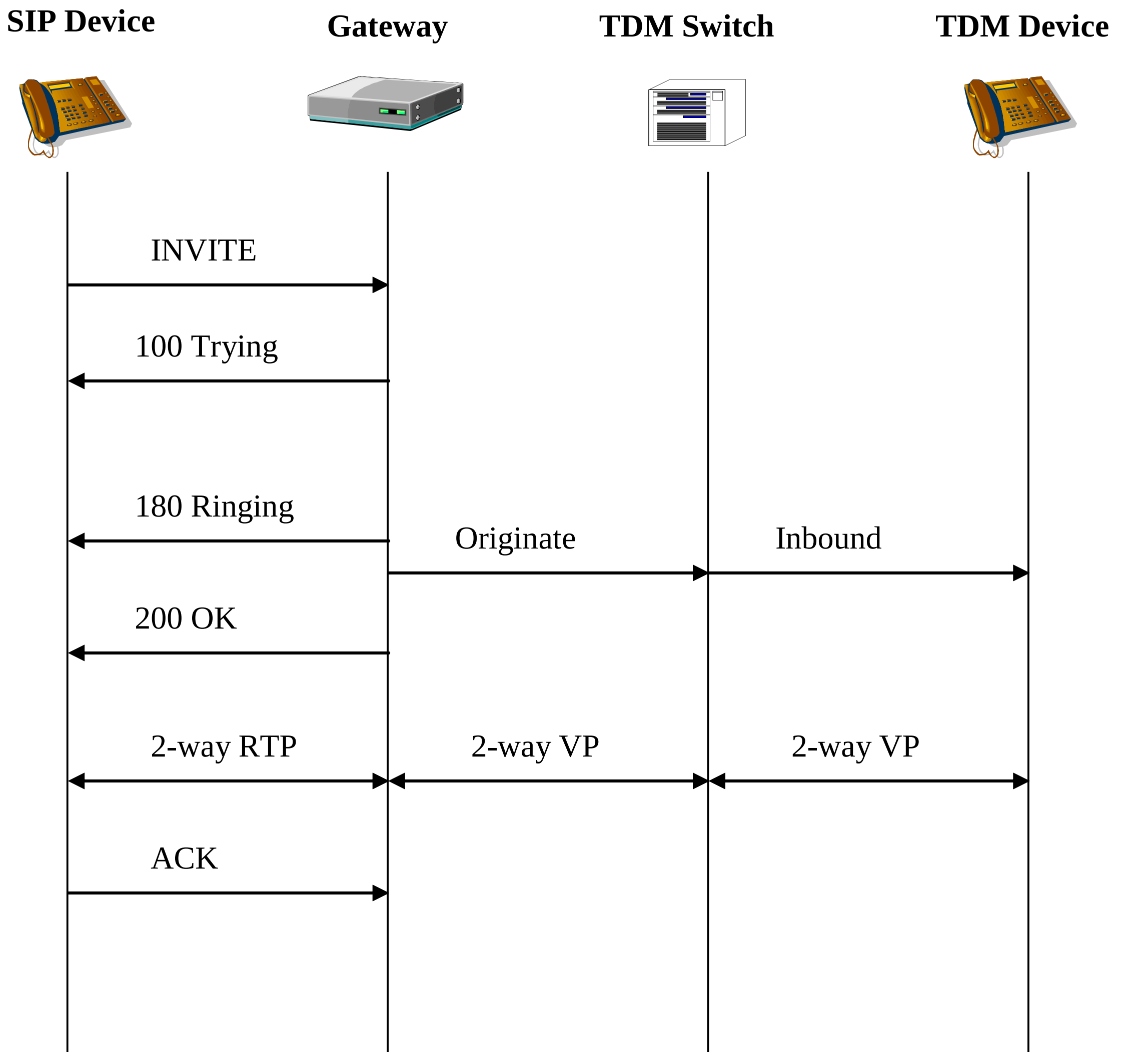
```
<----REGISTER sip:192.168.1.21:5060 SIP/2.0
From:sip:DialogicUser@DialogicGateway2000;tag=55B63246313536410000E8B5
To:sip:DialogicUser@DialogicGateway2000
Contact:<sip:DialogicUser@192.168.1.20:5060>
Expires:30
Call-ID:01B227235781400000000000@pbxgw.default.com
CSeq:1 REGISTER
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Via:SIP/2.0/UDP 192.168.1.20:5060;branch=z9hG4bK7F8D06F7386BC11230DA2CA7CE753381
Content-Length:0
```

11. Call Flows

The following diagrams show the message and control flows between the TDM network switch attached to the gateway and one or more SIP VOIP devices. A proxy/registrar/location server could be inserted between the gateway and the SIP devices, but their inclusion does not significantly change the call flows. For the sake of simplicity, a proxy/registrar/location server has been omitted from the diagrams.

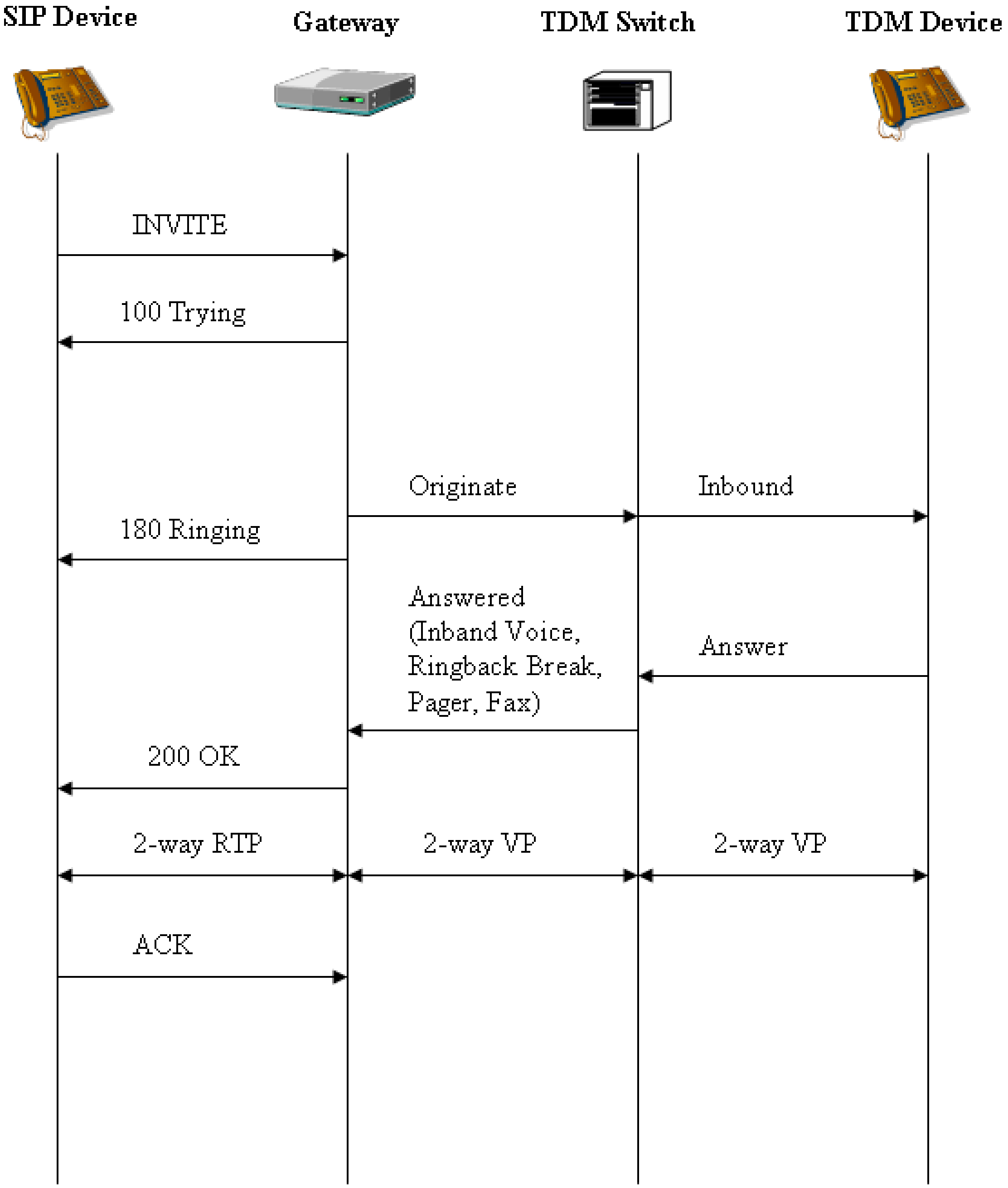
VOIP to TDM Success

The gateway generates 200 OK as soon as the TDM network line is seized and the destination number is dialed.



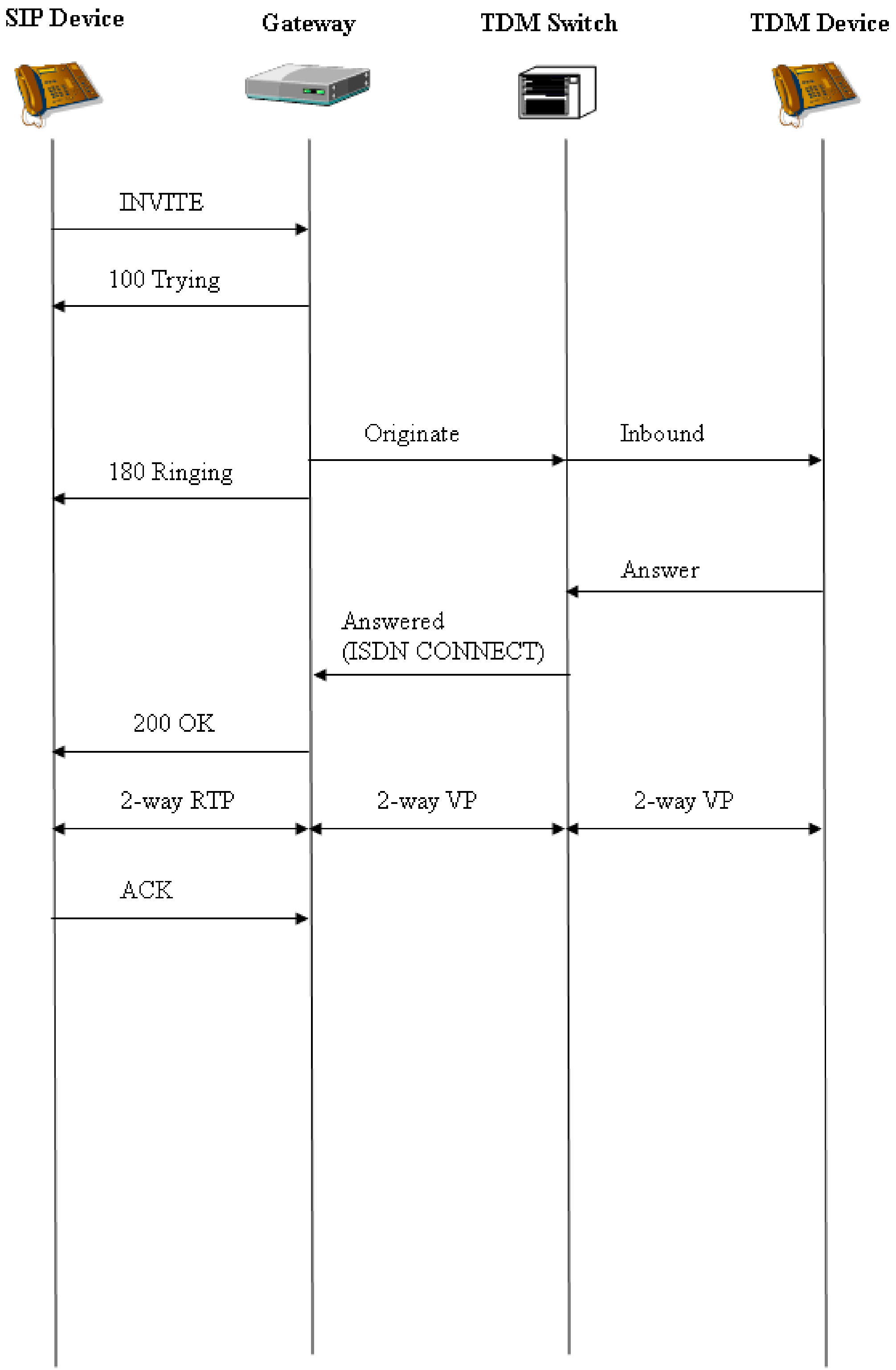
VOIP to TDM Success

The gateway generates 200 OK after a positive answer is detected. This may occur upon in-band voice activity detection, ring-back tone interruption, pager tone detection, or fax tone detection.



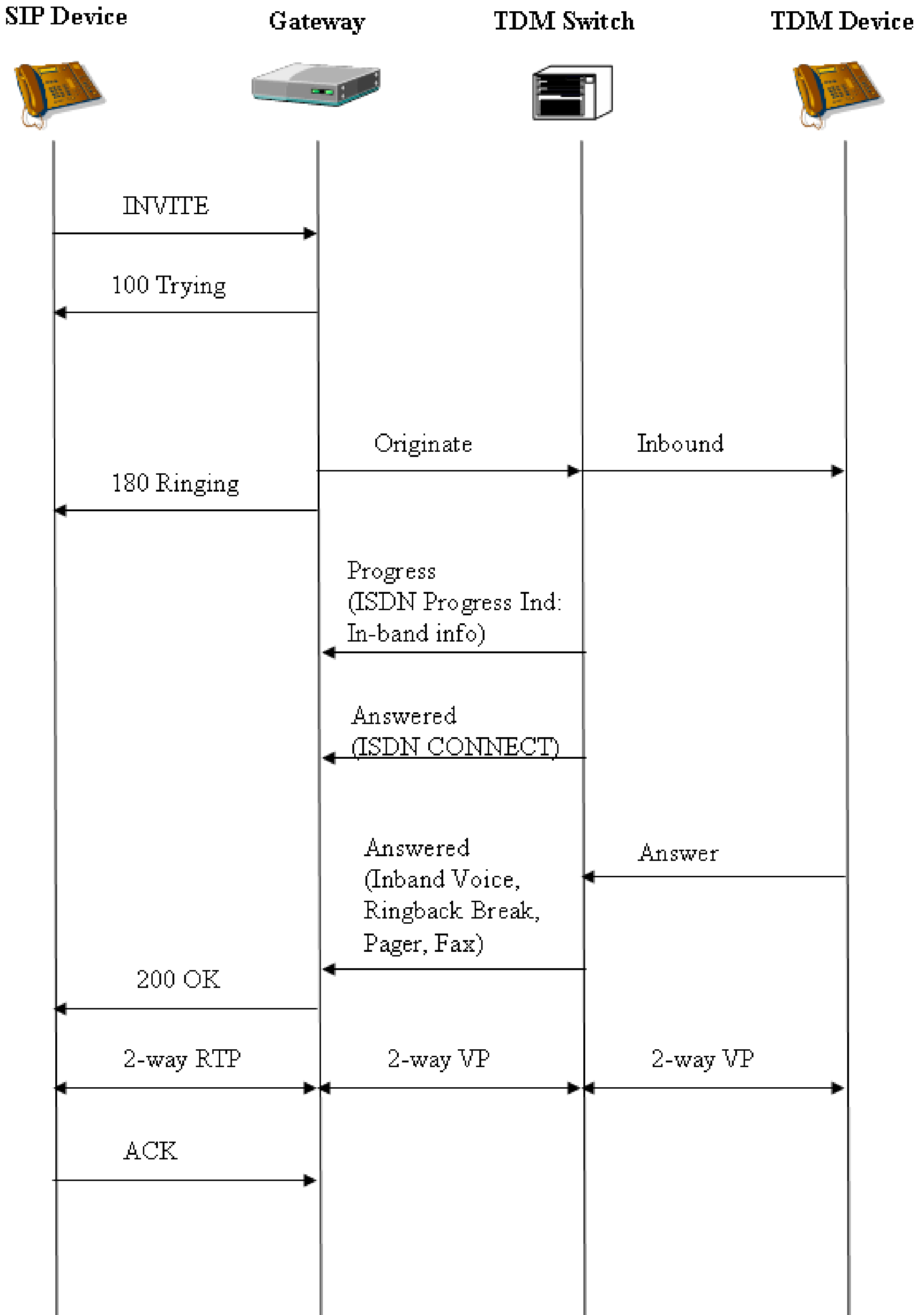
VOIP to TDM Success (ISDN CONNECT)

When the TDM interface uses an ISDN protocol, the gateway relies on the ISDN CONNECT message from the TDM Switch to determine when the call has been answered (exception to this is if the TDM switch sends an ISDN Progress Indication:(1 or 8), described in next section). The gateway generates 200 OK upon receiving the ISDN CONNECT message from the TDM Switch.



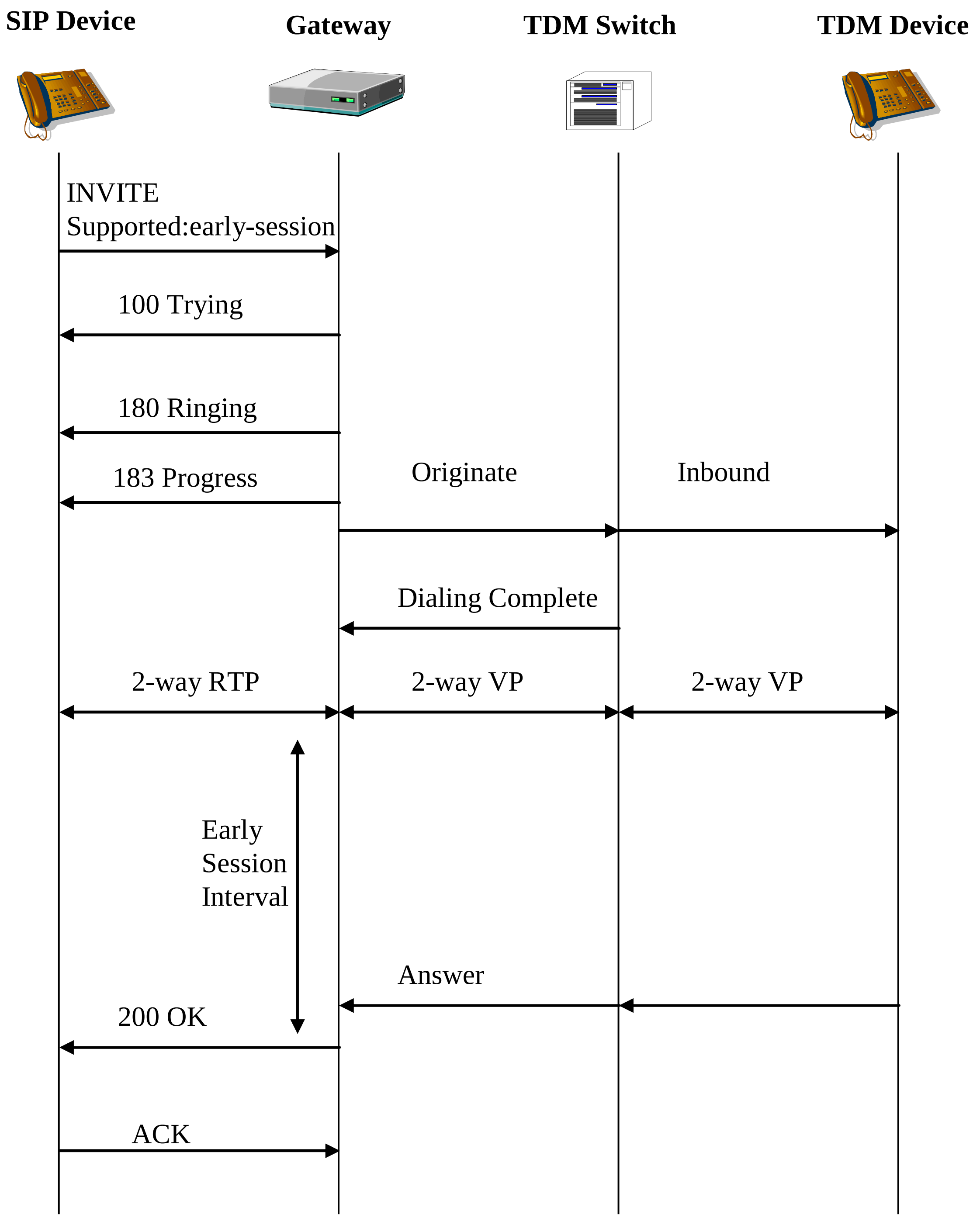
VOIP to TDM Success (ISDN Progress Indication:8 In-Band)

When the TDM interface uses an ISDN protocol, the gateway relies on the ISDN CONNECT message from the TDM Switch to determine when the call has been answered. However, if the TDM Switch determines that the call is not ISDN end-to-end, then it will send an ISDN Progress Indication:(1 or 8) message to the Gateway. An ISDN Progress Indication may be part of a SETUP, ALERTING, PROGRESS, or CONNECT message. This indication requests the Gateway to look in-band for call-progress tones since the ISDN CONNECT message cannot be relied upon to determine when the TDM endpoint has answered the call. In this situation, the Gateway will ignore the ISDN CONNECT message received from the TDM Switch and will rely on in-band call-progress tone detection to determine when the call has been answered. The Gateway generates 200 OK after a positive answer is detected. This may occur upon in-band voice activity detection, ring-back tone interruption, pager tone detection, or fax tone detection.



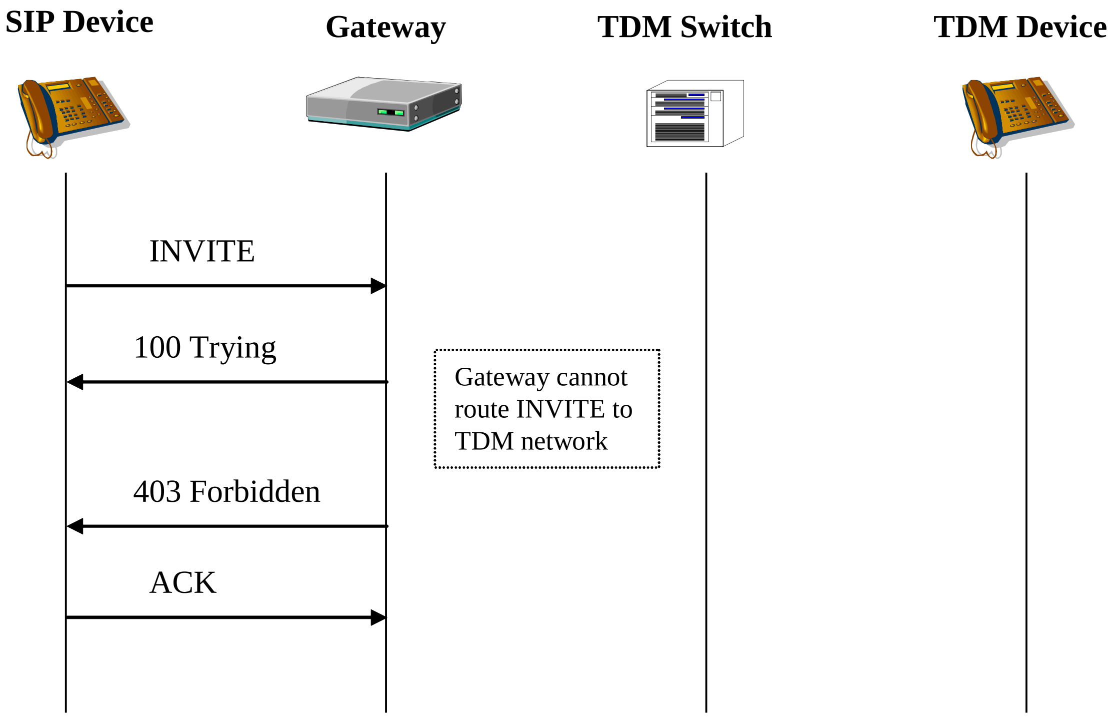
VOIP to TDM Success with Early Media

In this scenario, the SIP device requests for early media support using the 'Supported' header field with the 'early-session' option tag. The gateway responds with an additional 183 Session Progress message that indicates support for early media via the 'Content-Disposition' header and the 'early-session' option tag.



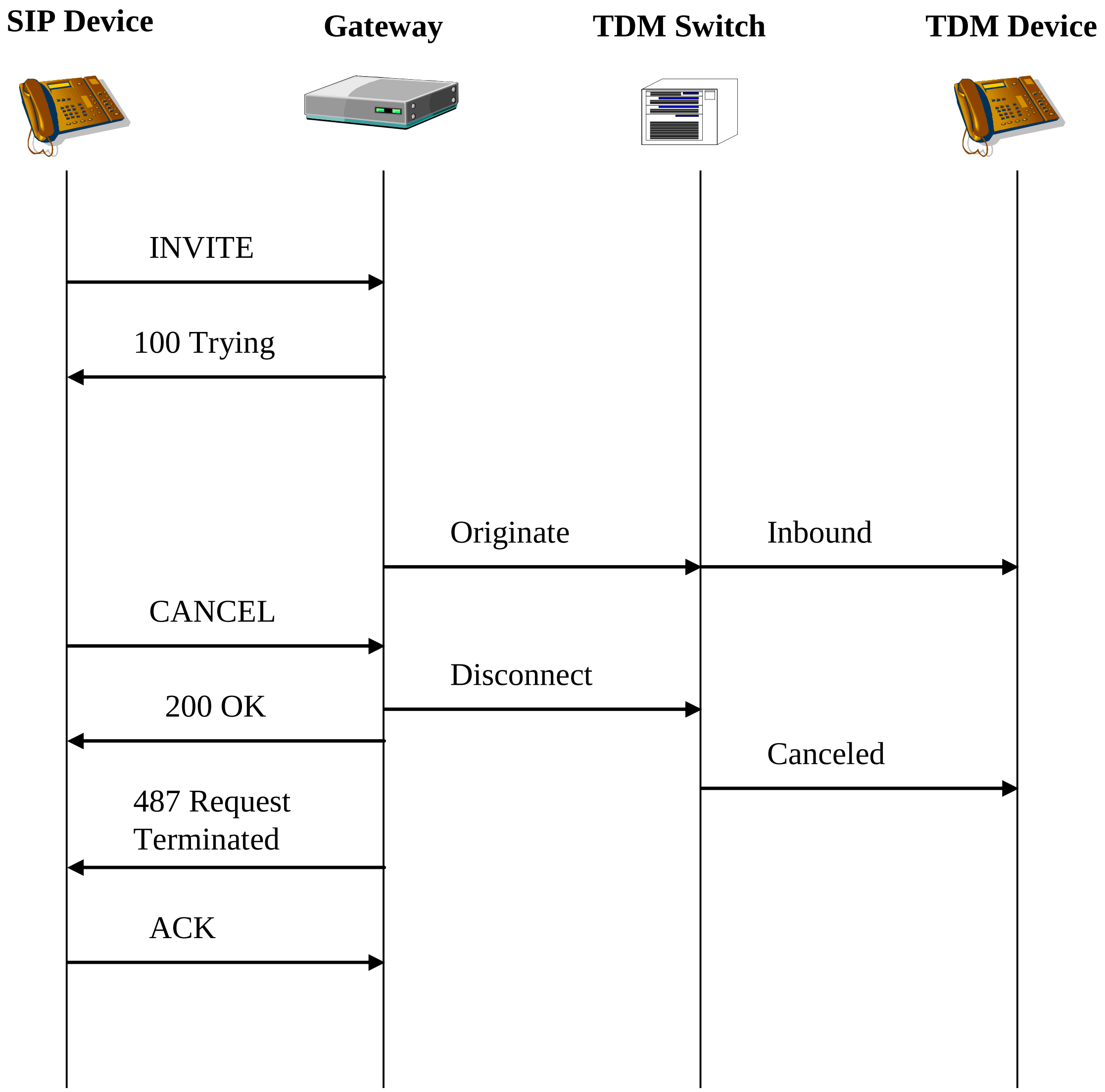
VOIP to TDM Failure – Call Un-Routable

If the INVITE request does not contain any valid destination number and no routing table rule may be positively matched, then the following scenario occurs.



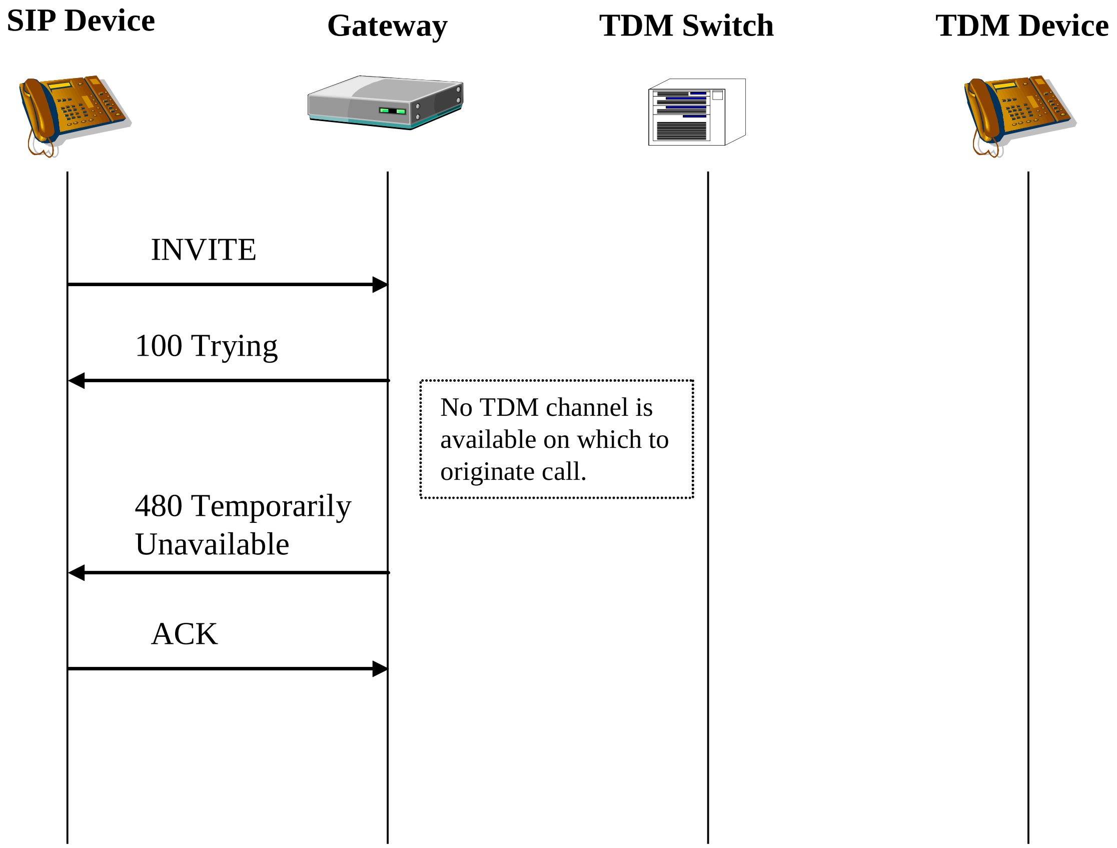
VOIP to TDM Failure – Call Canceled

If the INVITE request source cancels the request before the gateway completes the request, then the gateway will terminate the TDM network call.



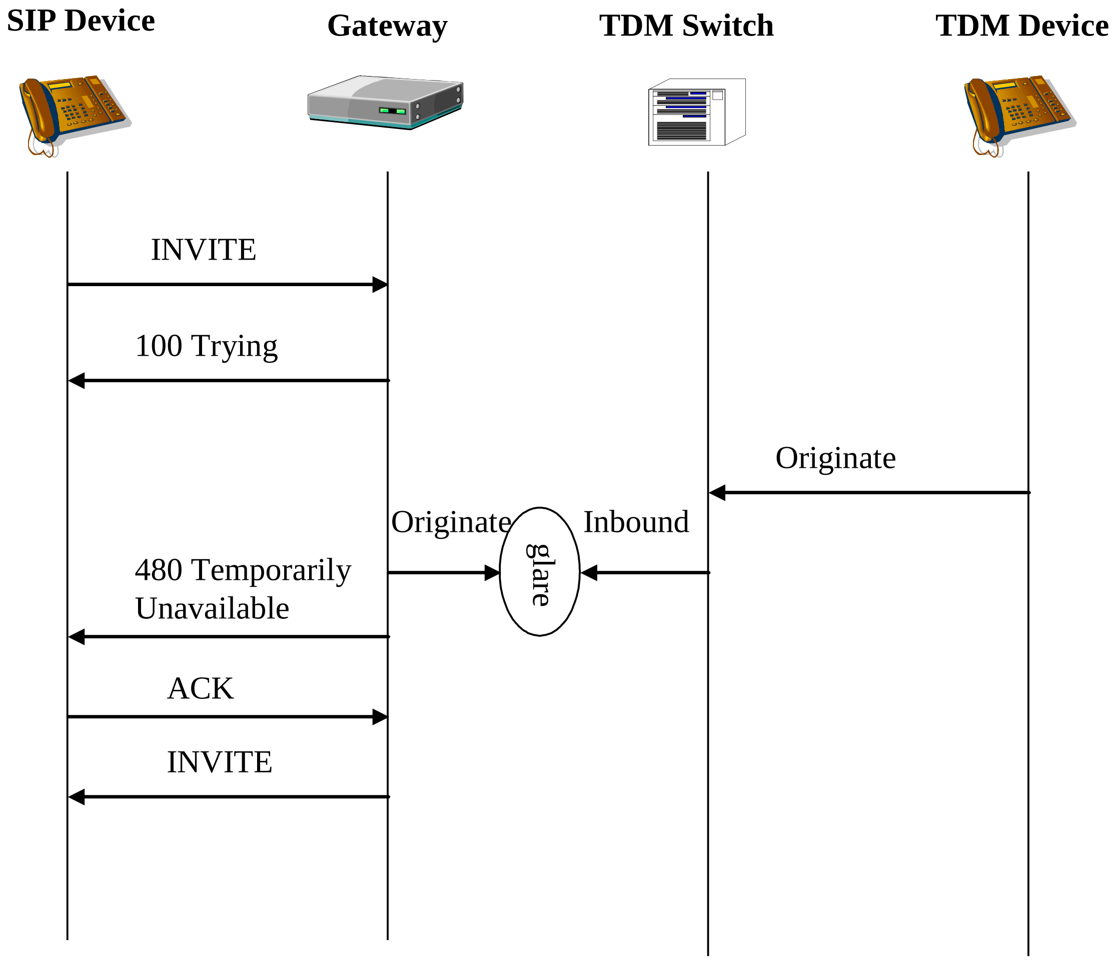
VOIP to TDM Failure – TDM Channel Unavailable

If there are no available gateway TDM channels, then the gateway responds with a 480.


SIP Device
Gateway
TDM Switch
TDM Device
INVITE
100 Trying
No TDM channel is available on which to originate call.
480 Temporarily Unavailable
ACK

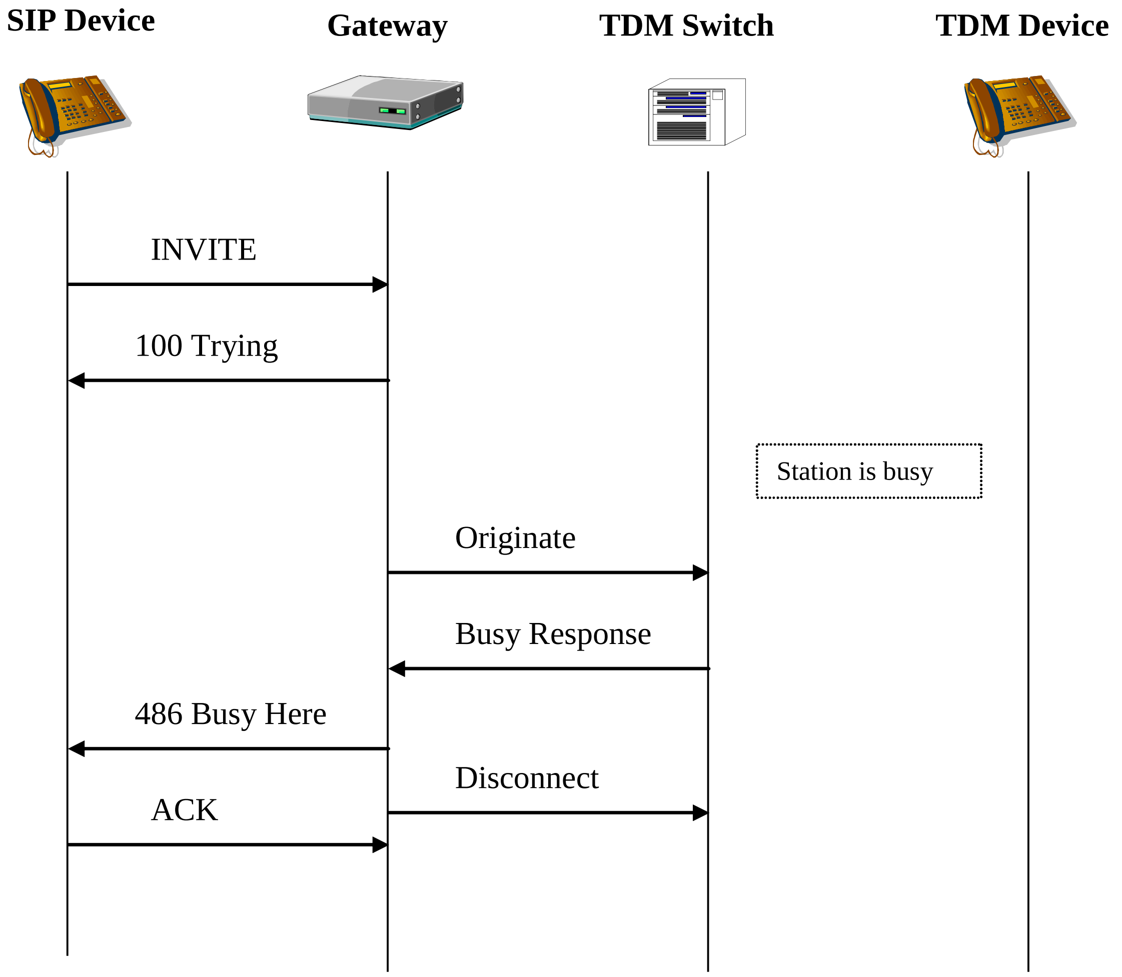
VOIP to TDM Failure – Glare

If a TDM network call is inbound on a TDM channel that is about to be used for a TDM call origination, then a glare condition occurs. The gateway fails the INVITE request, and presents a TDM-To-VOIP INVITE request for the inbound TDM call.



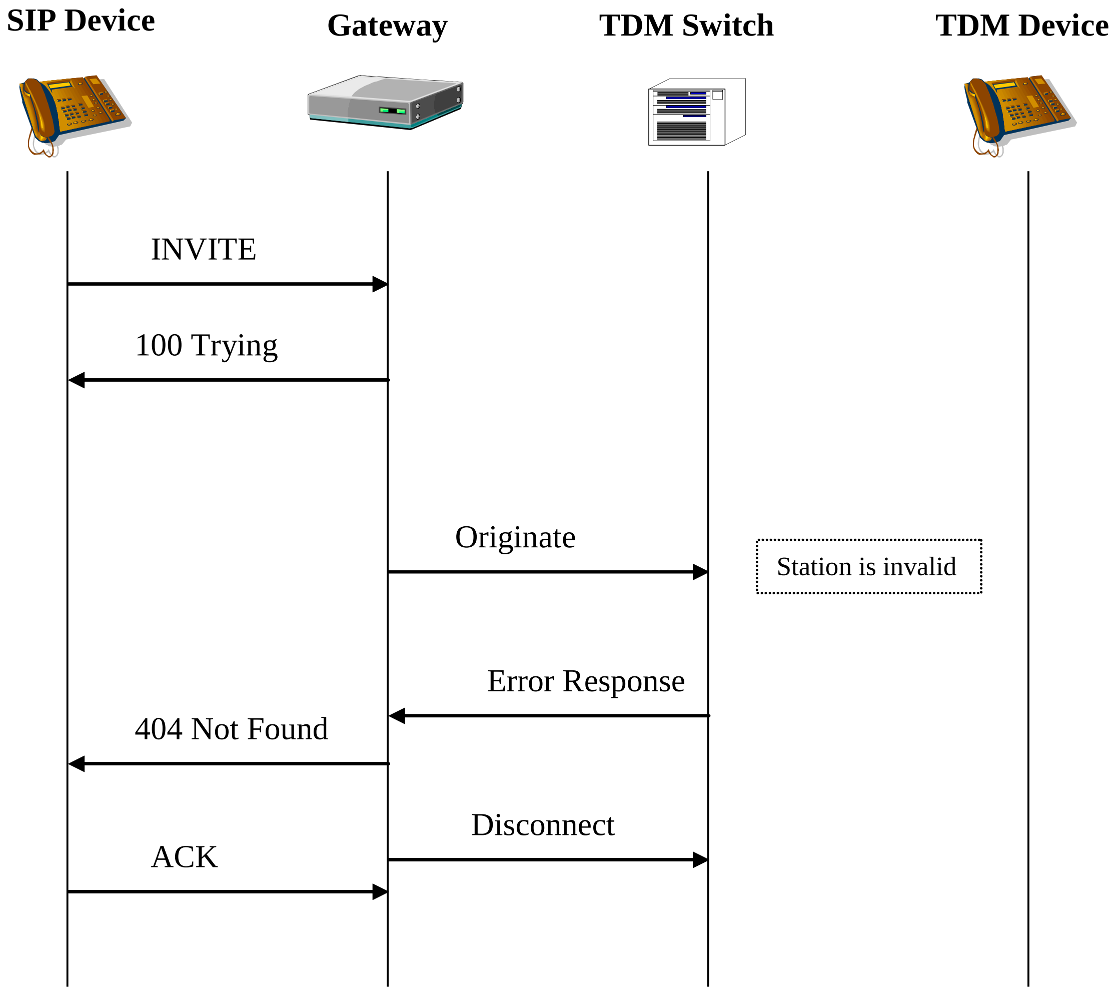
VOIP to TDM Failure – Busy Response

A busy response detected during TDM call origination results in a failure response to the INVITE request and the TDM call is terminated. The busy response could be a busy-tone or an ISDN busy-disconnect.



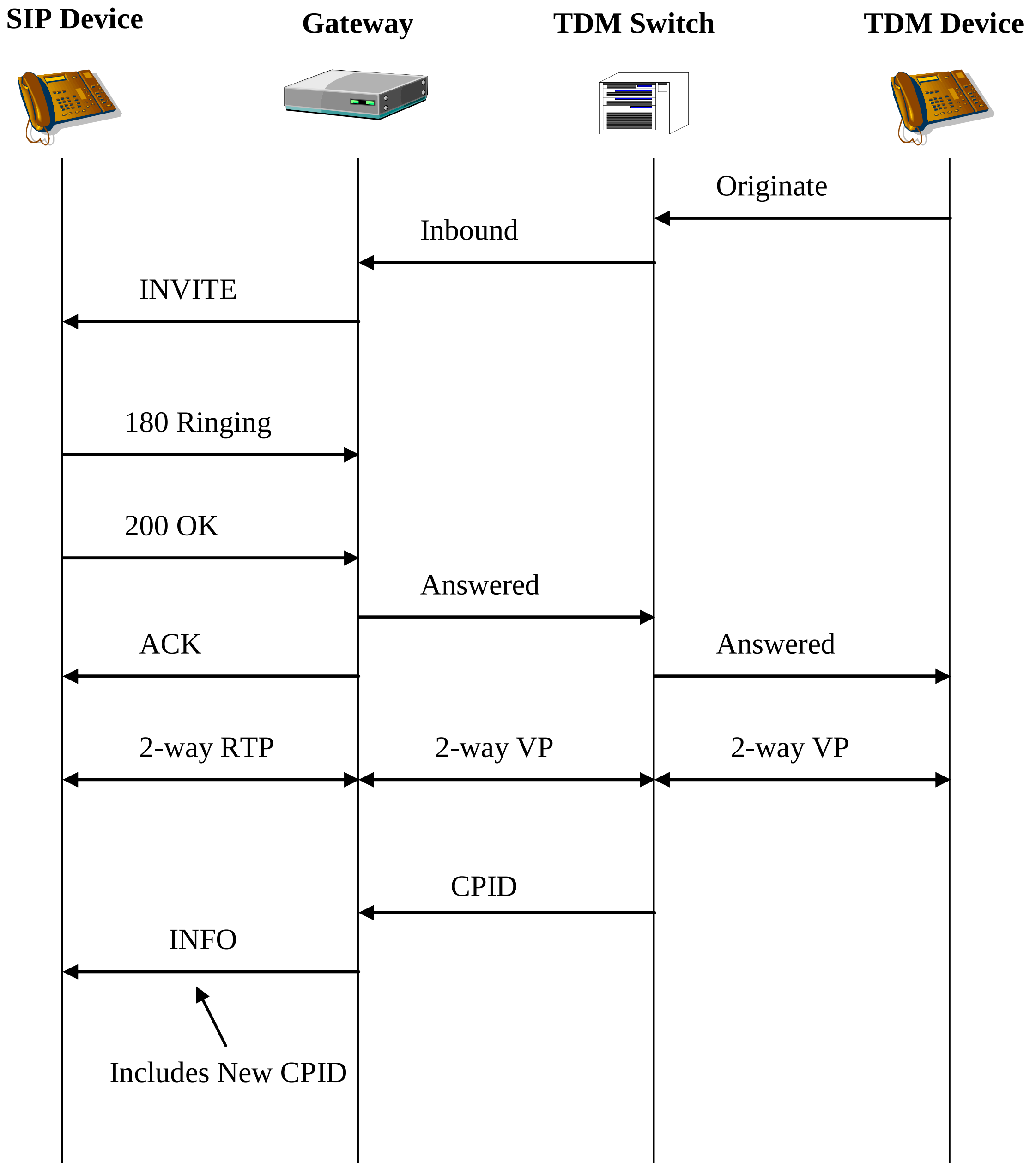
VOIP to TDM Failure – Error Response

An error response detected during TDM call origination results in a failure response to the INVITE request and the TDM call is terminated. The error response could be an error tone or an ISDN error-disconnect.



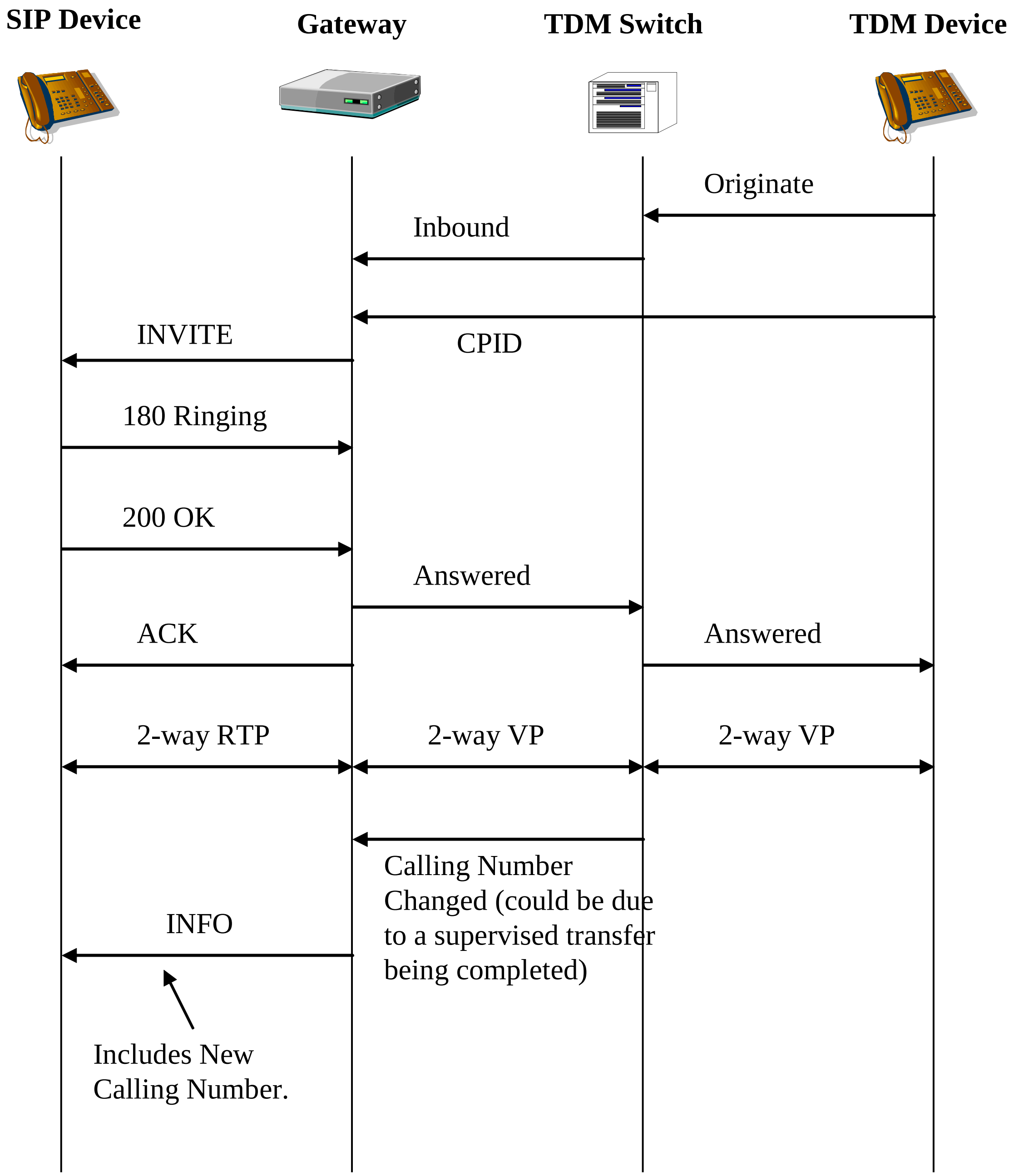
TDM to VOIP Success (Post-INVITE CPID)

The INFO method is generated only if the TDM call party information was not available at the time of the INVITE request generation, but has now become available. This can occur as a result of Type II CPID, in which the CPID is available only after the TDM call is answered.



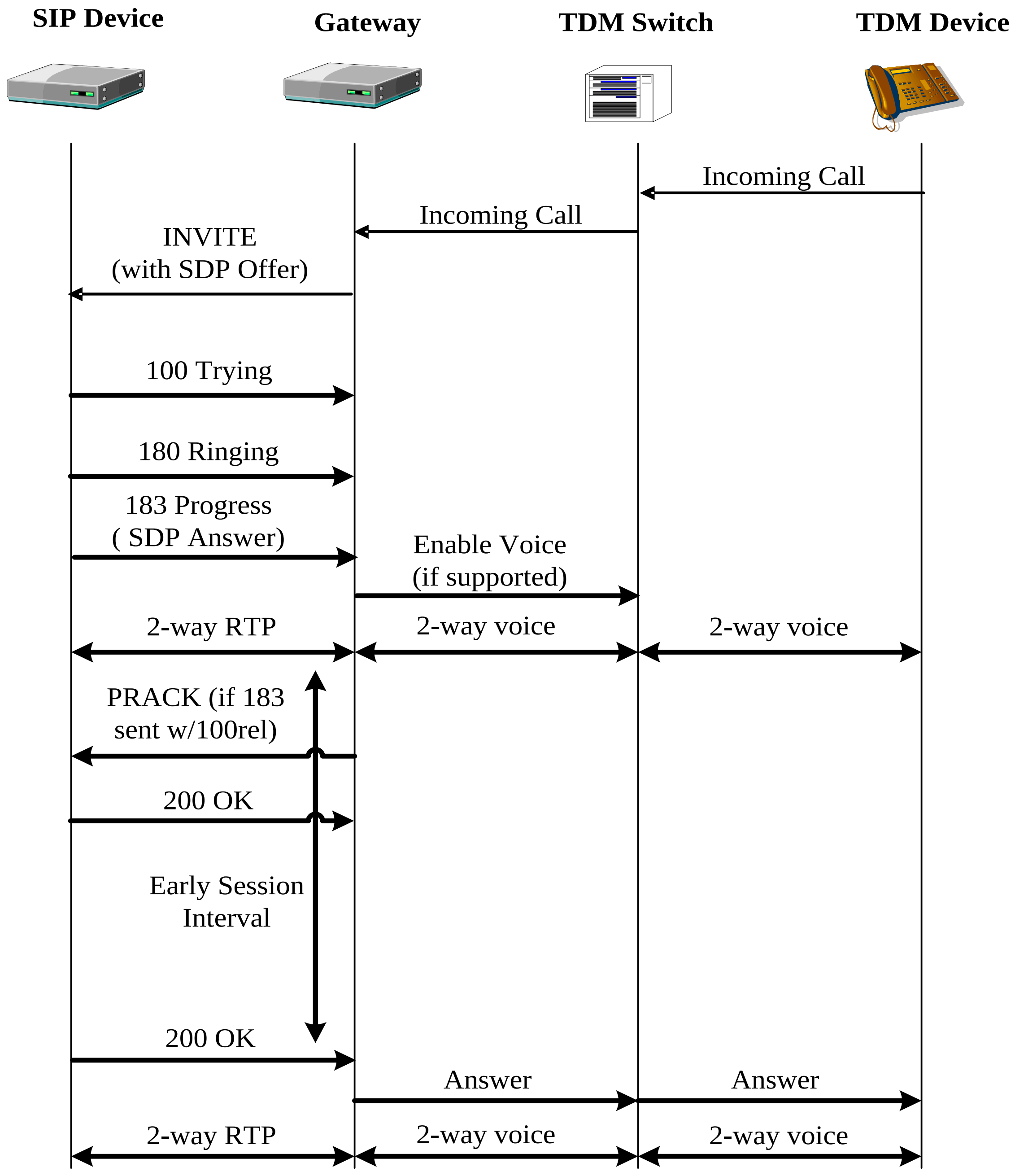
TDM to VOIP Success (Calling Number Updated)

If the calling number is updated either before or after the call is answered (either ringing or connected state), the Media Gateway can send an INFO message update to the VOIP endpoint.



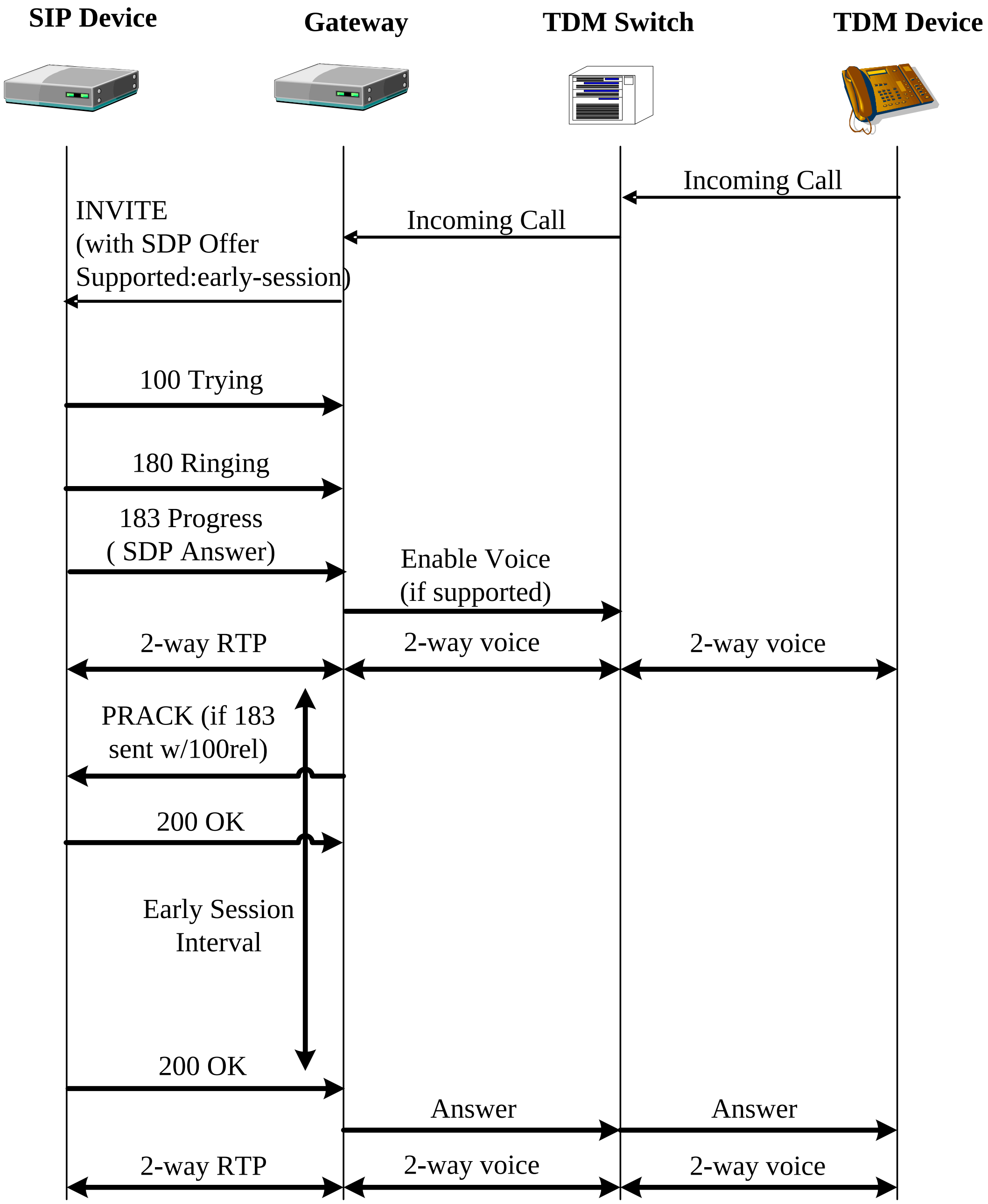
TDM to VOIP Success with Early Media (Gateway Model)

In this scenario, the SIP device requests for early media support by responding to the initial INVITE sent by the gateway – which included a SDP Offer – with a SDP Answer in a 183 Session Progress message. Note that the "Gateway" model is enabled by setting the gateway's **sipEarlyMediaSupport** parameter to 'Always'.



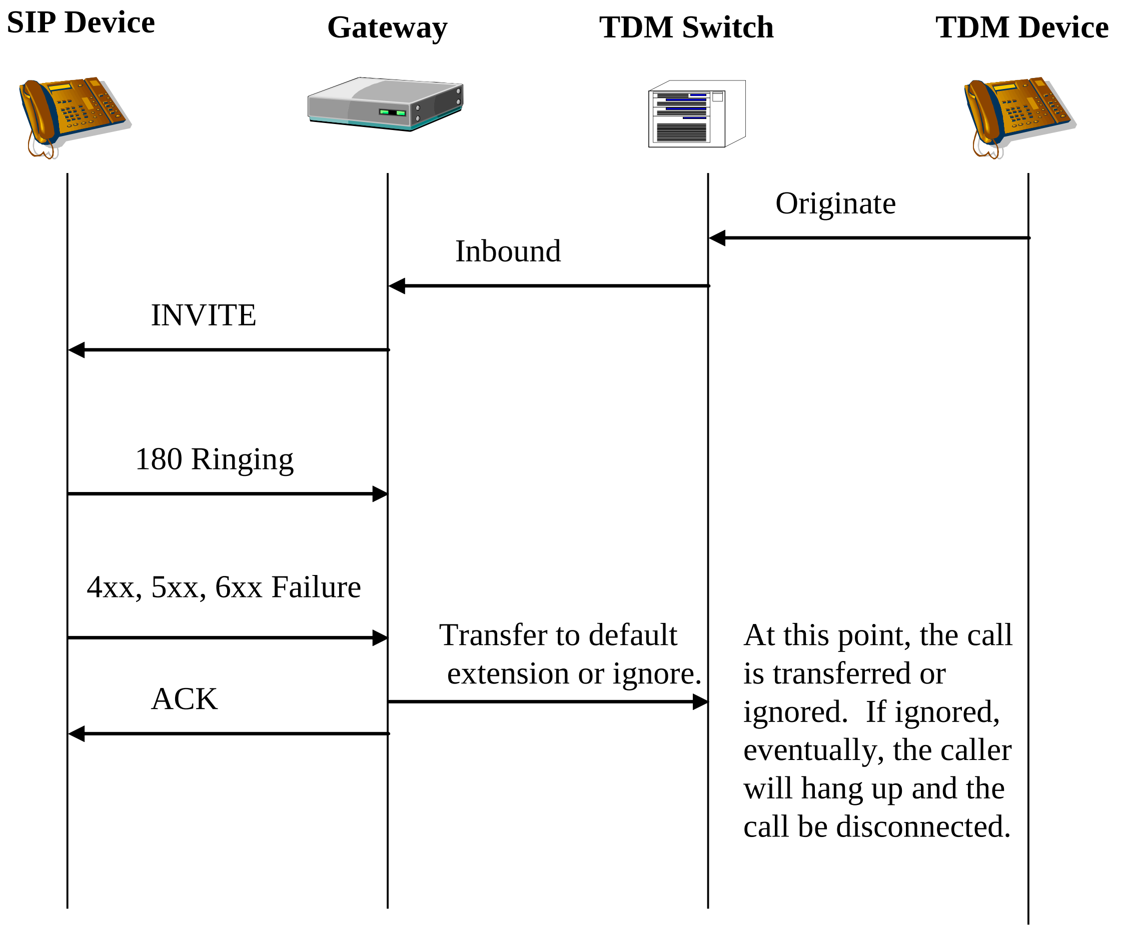
TDM to VOIP Success with Early Media (Application Model)

In this scenario, the SIP device requests for early media support by responding to the initial INVITE sent by the gateway – which included a Session header set to 'early-session and a SDP Offer – with a SDP Answer in a 183 Session Progress message. Note that the "Application" model is enabled by setting the gateway's **sipEarlyMediaSupport** parameter to 'On-Demand'.



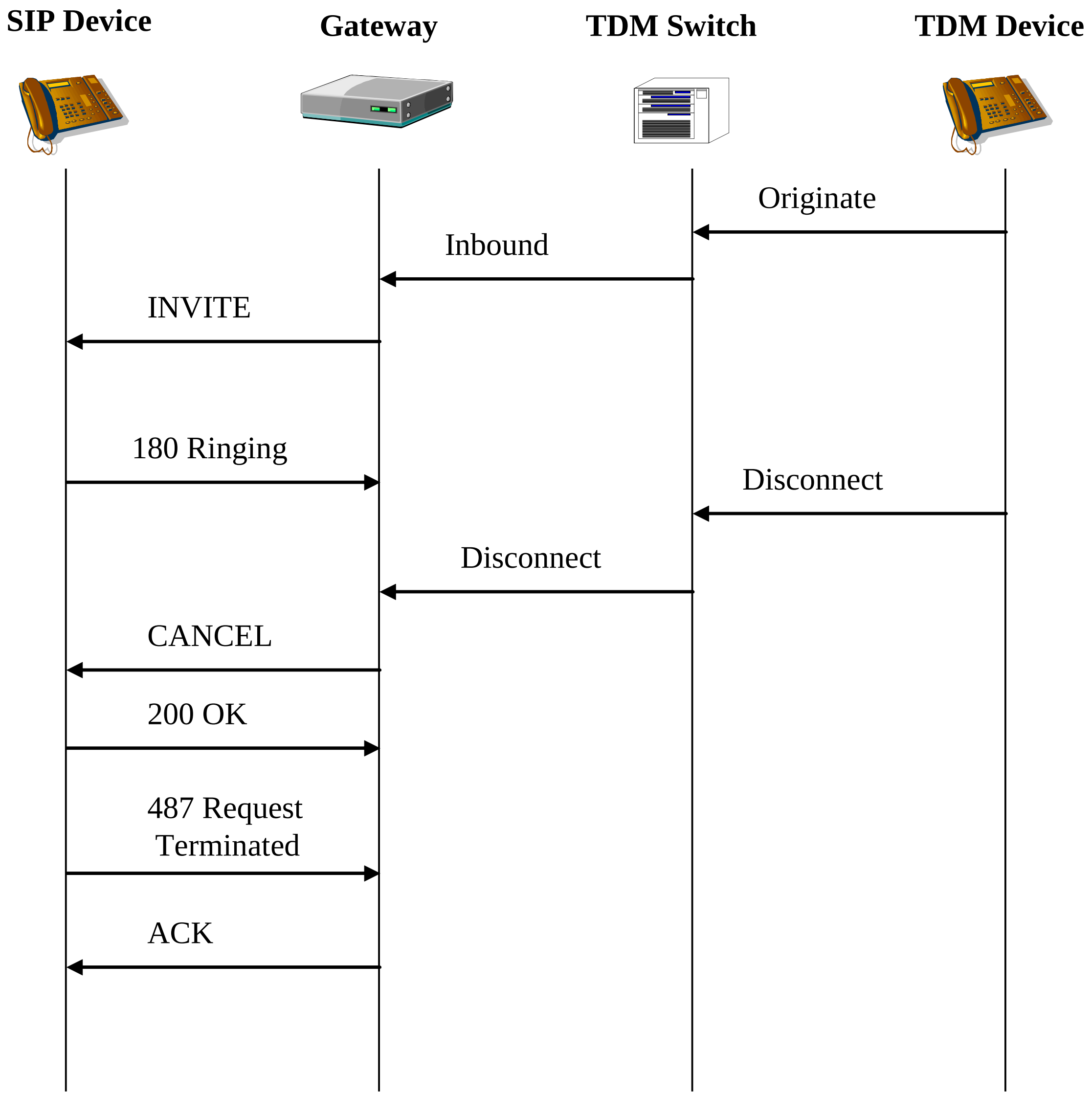
TDM to VOIP Failure - Rejected

If the VOIP destination rejects the call with any error response, or the INVITE request times out, then the TDM call is transferred to the default TDM network destination, or the call is ignored. If a default TDM network destination is configured, then the TDM call is answered and immediately - transferred (unsupervised) to the default destination. If no default TDM network destination is configured, then the TDM call is ignored.


SIP Device
Gateway
TDM Switch
TDM Device
Originate
Inbound
INVITE
180 Ringing
4xx, 5xx, 6xx Failure
Transfer to default extension or ignore.
ACK
At this point, the call is transferred or ignored. If ignored, eventually, the caller will hang up and the call be disconnected.

TDM to VOIP Failure - TDM Cancel

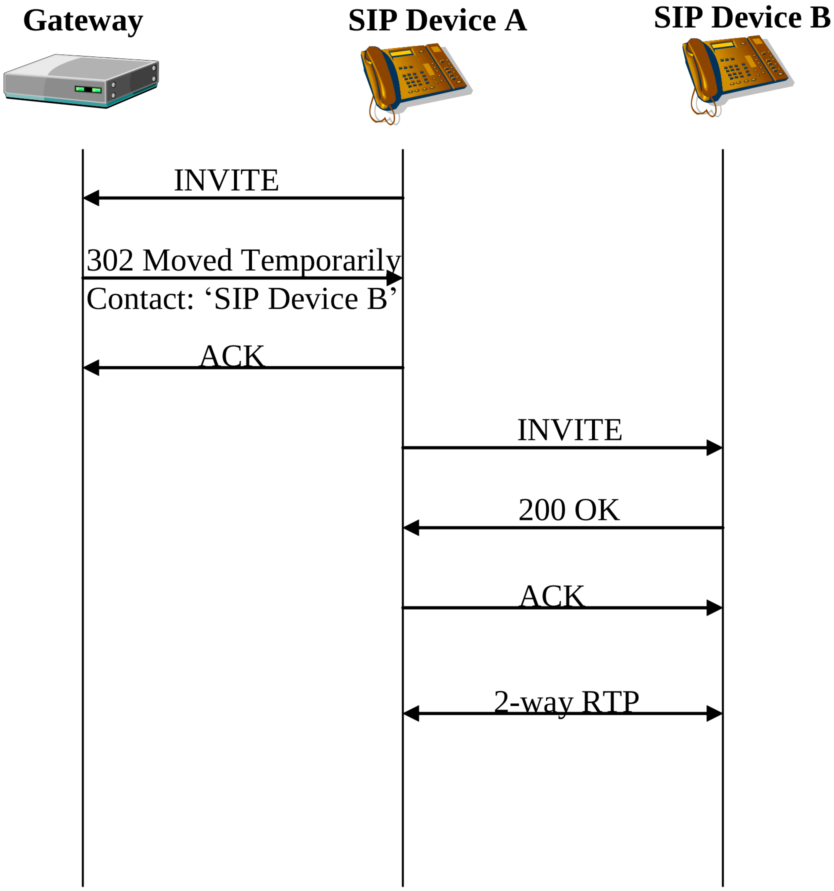
If the TDM network call terminates before the INVITE request is answered, then the gateway will cancel the INVITE request.



VOIP to VOIP Success – Redirect Routing

The gateway supports the ability to perform VOIP to VOIP Redirect Routing. In this case, the gateway simply responds to an incoming INVITE from a SIP peer with a 302 response. This call flow does not show any 1xx messages for the purpose of simplification.

The case shown here is one in which SIP Device A sends an INVITE to the gateway. Due to a routing table configuration which indicates Redirect as opposed to Bridged routing, the gateway responds to SIP Device A with a 302 response. Inside the 302 response is a 'Contact' header which contains the URI of SIP Device B. At this time, SIP Device A may redirect its INVITE to SIP Device B if it chooses to do so. In this scenario, no media is exchanged with the gateway and a call is connected between the external SIP devices with no further gateway interaction.

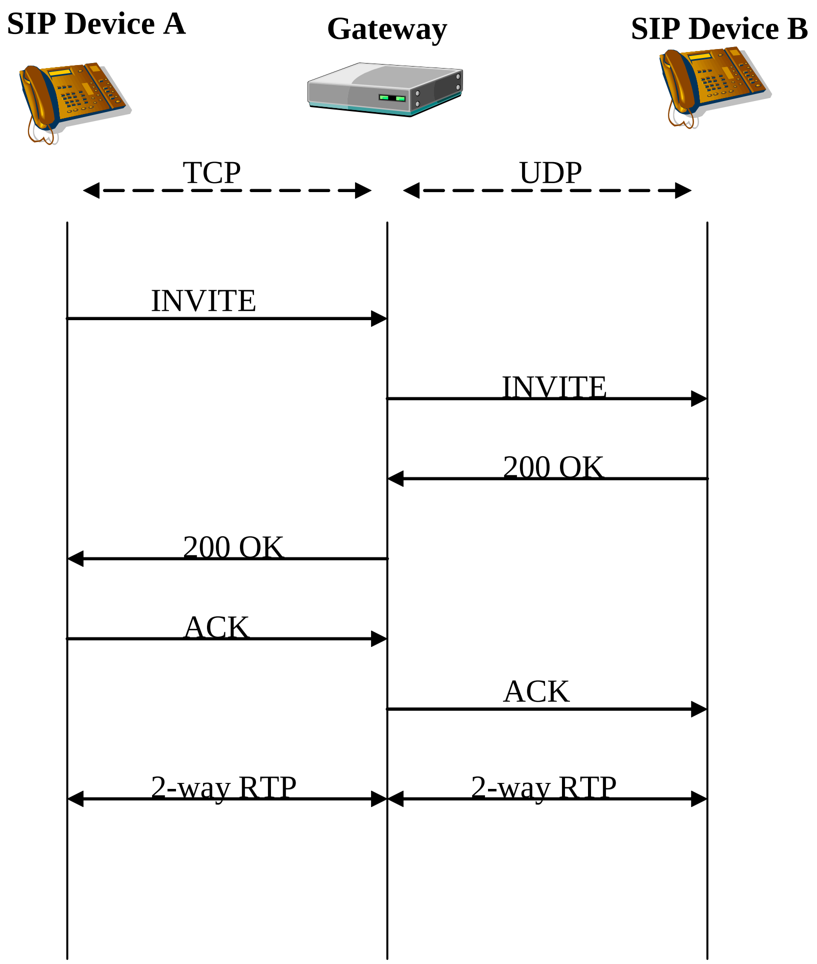


VOIP to VOIP Success – Bridged Routing with Pass Through

The gateway supports the ability to perform VOIP to VOIP Bridged Routing. In this case, the gateway bridges two disparate signaling networks or devices. This call flow does not show any 1xx messages for the purpose of simplification.

The case shown here is one in which SIP Device A is configured to communicate to the gateway via TCP. SIP Device B may be a SIP endpoint which is only capable of SIP signaling using UDP and is not capable of TCP communication. The gateway solves this problem by bridging the signaling at the transport layer. All signaling communication between SIP Device A and the gateway takes place via TCP. Due to the routing table configuration, the gateway decides to route the incoming call to SIP Device B which is configured for UDP. The gateway provides the transport layer translation and the SIP signaling is sent to SIP Device B via UDP.

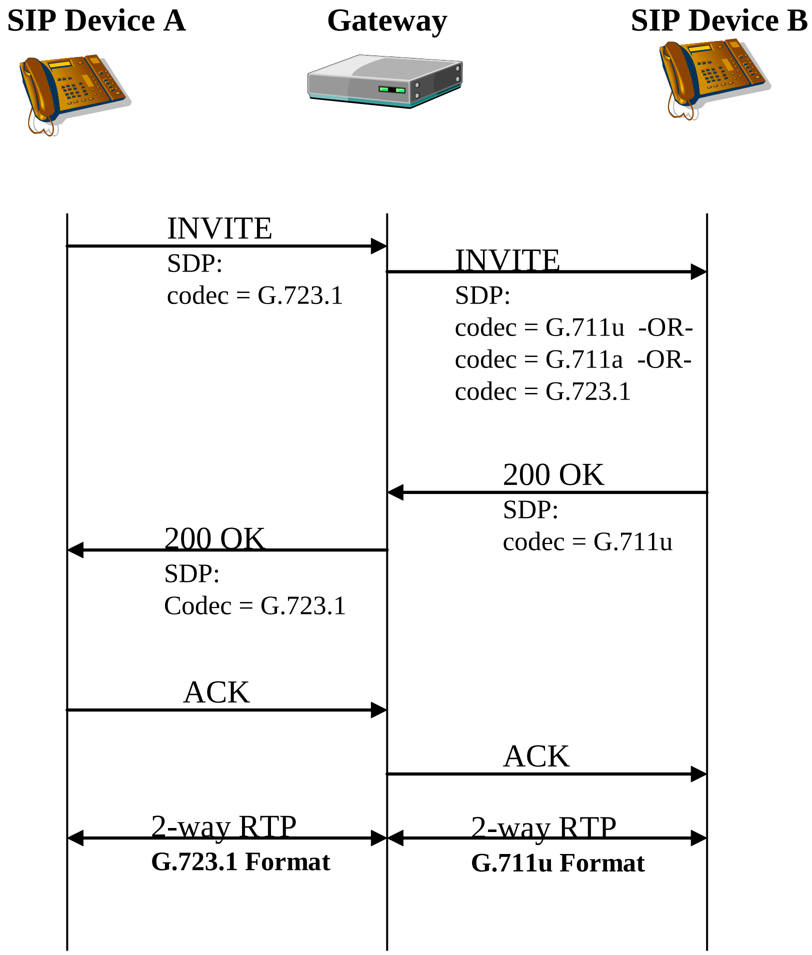
Once the call is established, the media passes through the gateway. SIP Device A sends its media to the gateway. The gateway in turn receives the media from SIP Device A and passes it through to SIP Device B. Media also flows in the opposite direction from SIP Device B through the gateway and on to SIP Device A. In this scenario, there is no transcoding or other modification to the media. All signaling and media originating from SIP Device A and SIP Device B pass through the gateway with only modification to the SIP signaling.



The VOIP to VOIP routing aspect of the gateway supports many of the same bridged routing call flows which are seen with VOIP to TDM and TDM to VOIP routing. This includes but is not limited to Re-INVITE, messages such as MWI and CPID, transfers involving the REFER method and FAX support . These call flows and message flows are illustrated for VOIP to TDM in other parts of this document.

VOIP to VOIP Success – Bridged Routing with Transcoding

Transcoding is the ability of the gateway to transform RTP packets from one codec format to a different codec format. Transcoding is employed in VOIP to VOIP bridged routing when SIP Device A and SIP Device B would prefer to communicate with the gateway via differing codec formats.

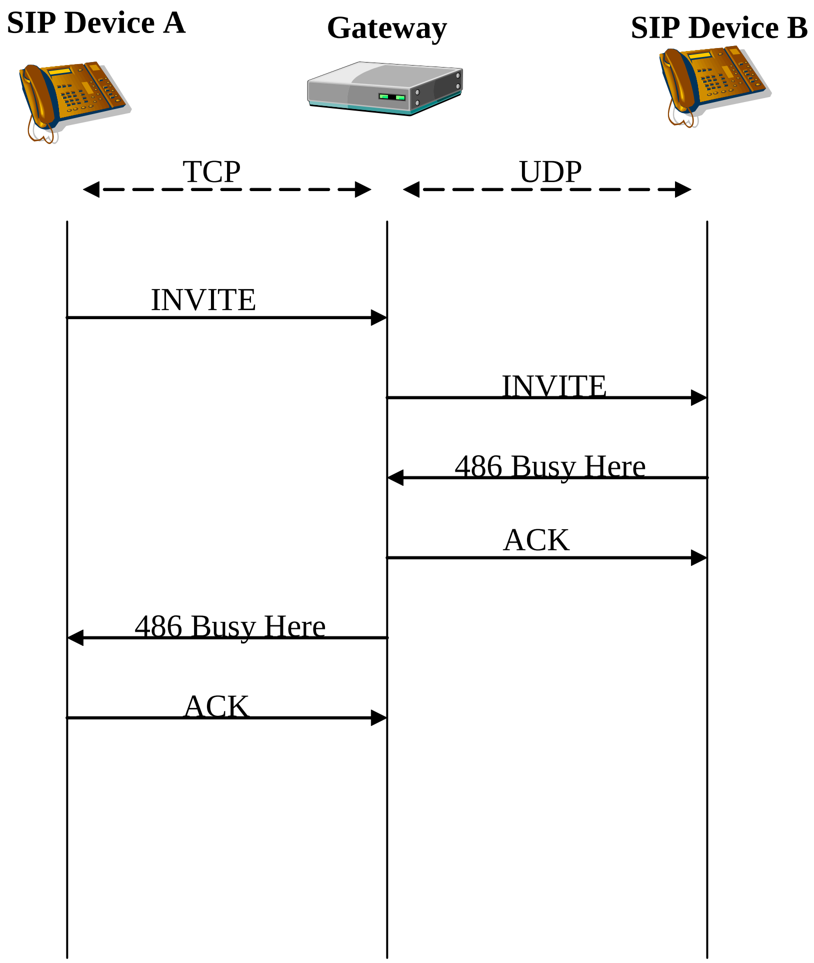


For example, it is possible that SIP Device A sends the INVITE to the Gateway with an SDP that allows only G.723.1 format. When the gateway builds the corresponding INVITE to be sent to SIP Device B, the gateway will build and send SDP format that offers all supported codecs as specified by the chosen Network Group. In this case an INVITE was sent with an SDP codec offer allowing G.711u, G.711a and G.723.1 (this is completely determined by the Network Group selection and is independent of the SDP in the INVITE from SIP Device A).

Suppose that SIP Device B selects the G.711u codec. SIP Device B will place this selection in the SDP of the 200 OK response sent back to the gateway. The gateway will then take care of transcoding the received RTP packets. The G.711u packets from SIP Device B will be transcoded to G.723.1 format before re-sending the packet to SIP Device A. Likewise, the G.723.1 RTP packets received from SIP device A will be transcoded to G.711u before re-sending the packet to SIP Device B.

VOIP to VOIP Failure – Bridged Routing with Pass Through

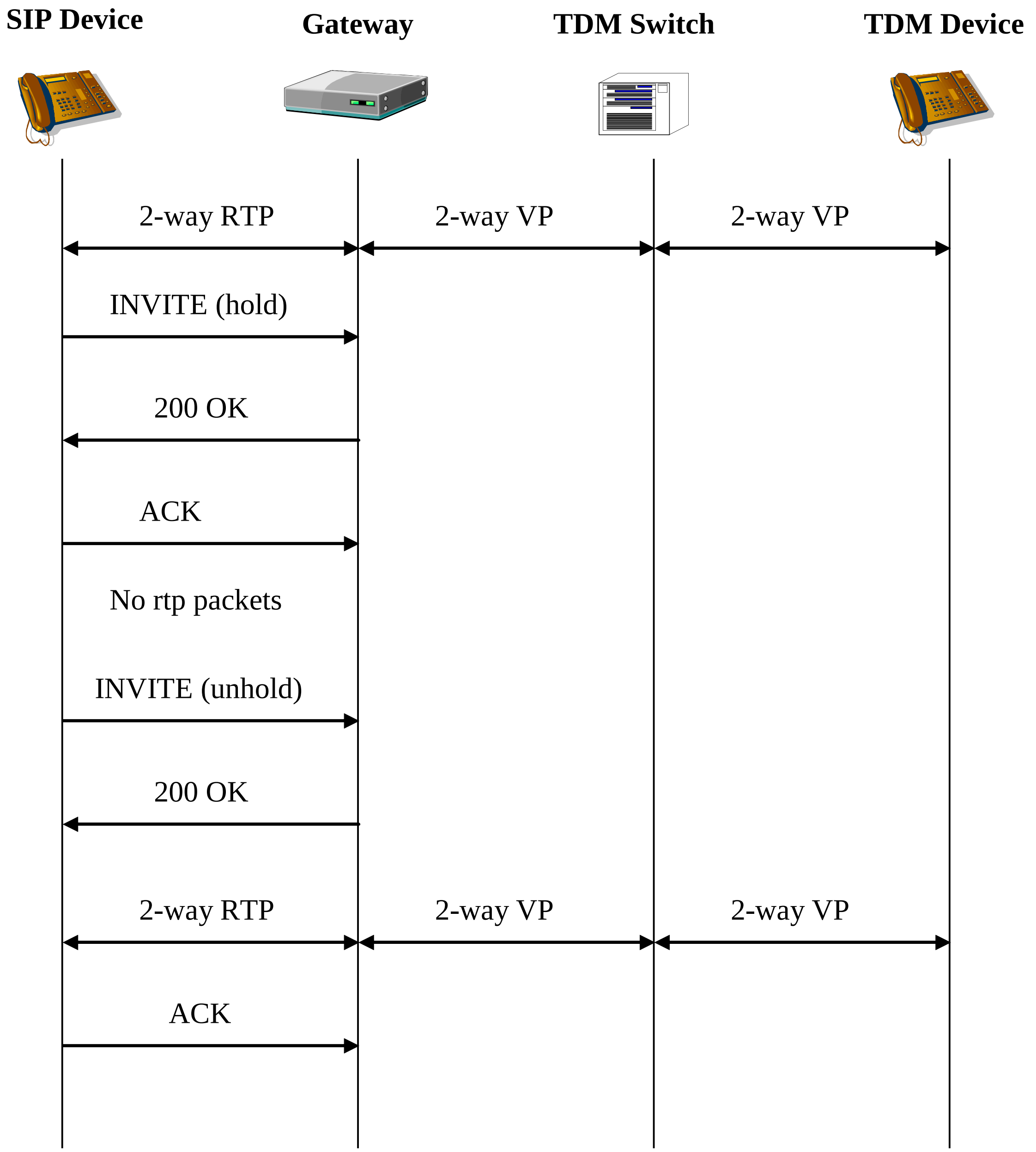
This scenario illustrates a failure in a VOIP to VOIP routing due to SIP Device B rejecting the INVITE. This case is a rejection due to a Busy SIP peer although the reason for the failure may be any one of the valid SIP error codes.



Hold, Unhold

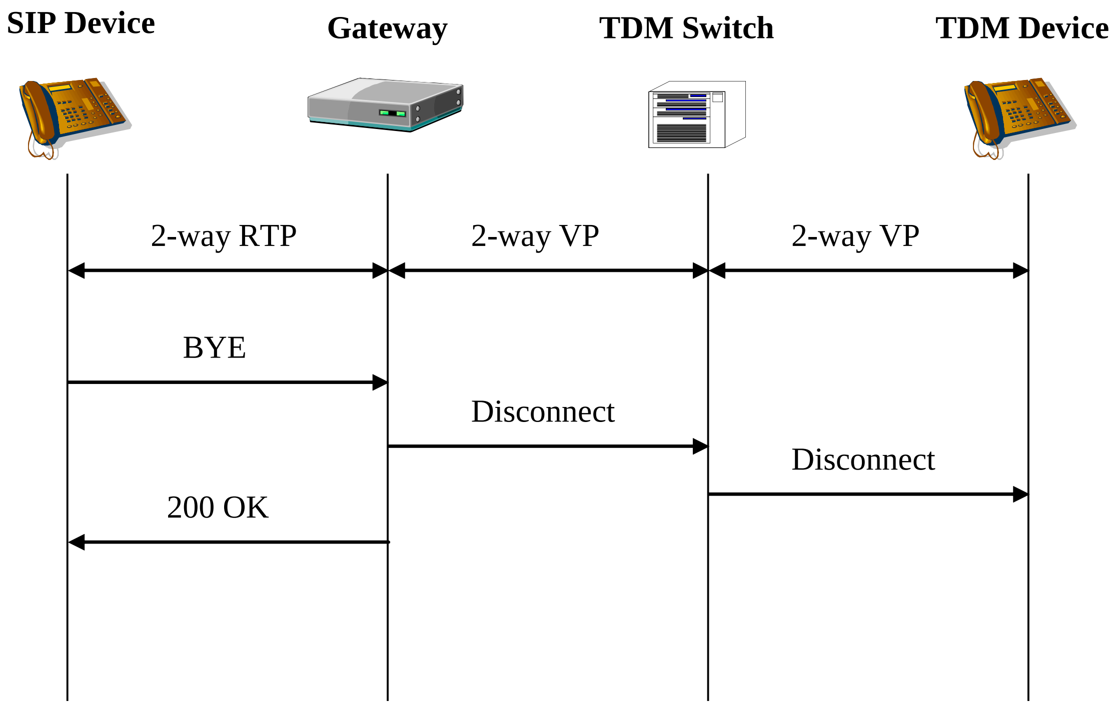
Section 0 describes how a VoIP call is placed on-hold and subsequently released from hold.

When a VoIP call on the Gateway is placed on-hold, the state of the TDM network call is unchanged. Note: the TDM network call is NOT put on-hold since a transfer request may be made to the 'on-hold' call, at which point the TDM call must not be in a held state. See the transfer call flows.



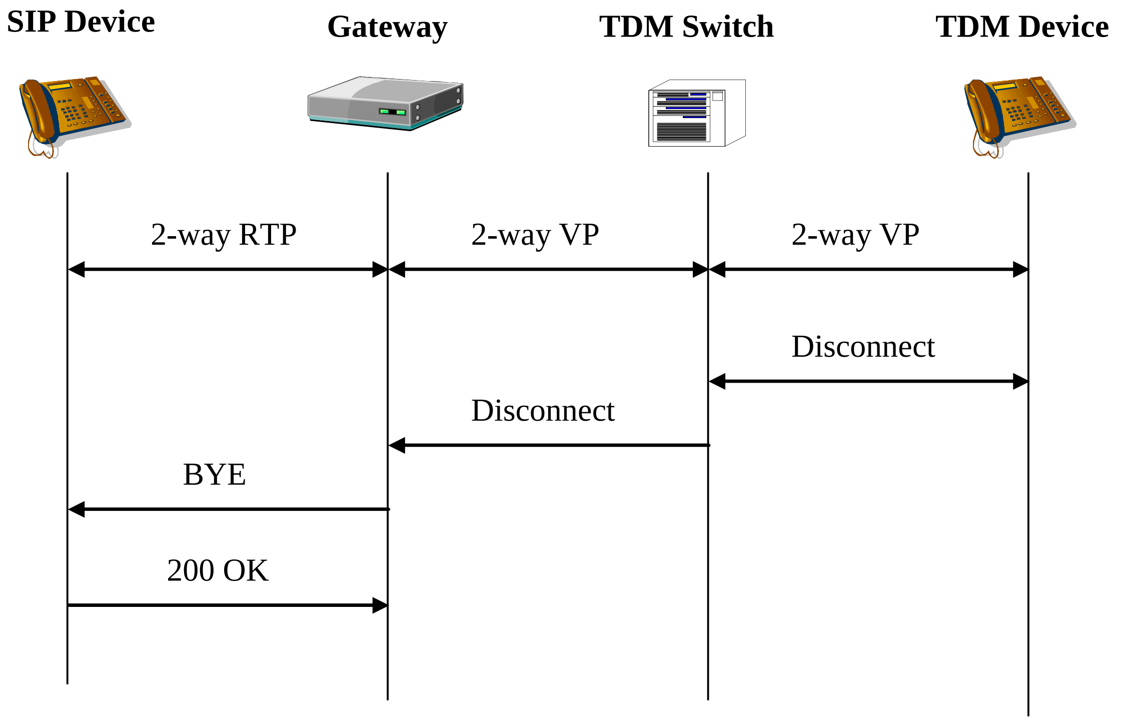
VOIP Call Drop

As soon as a BYE request is received from the VOIP device, the TDM call is terminated.



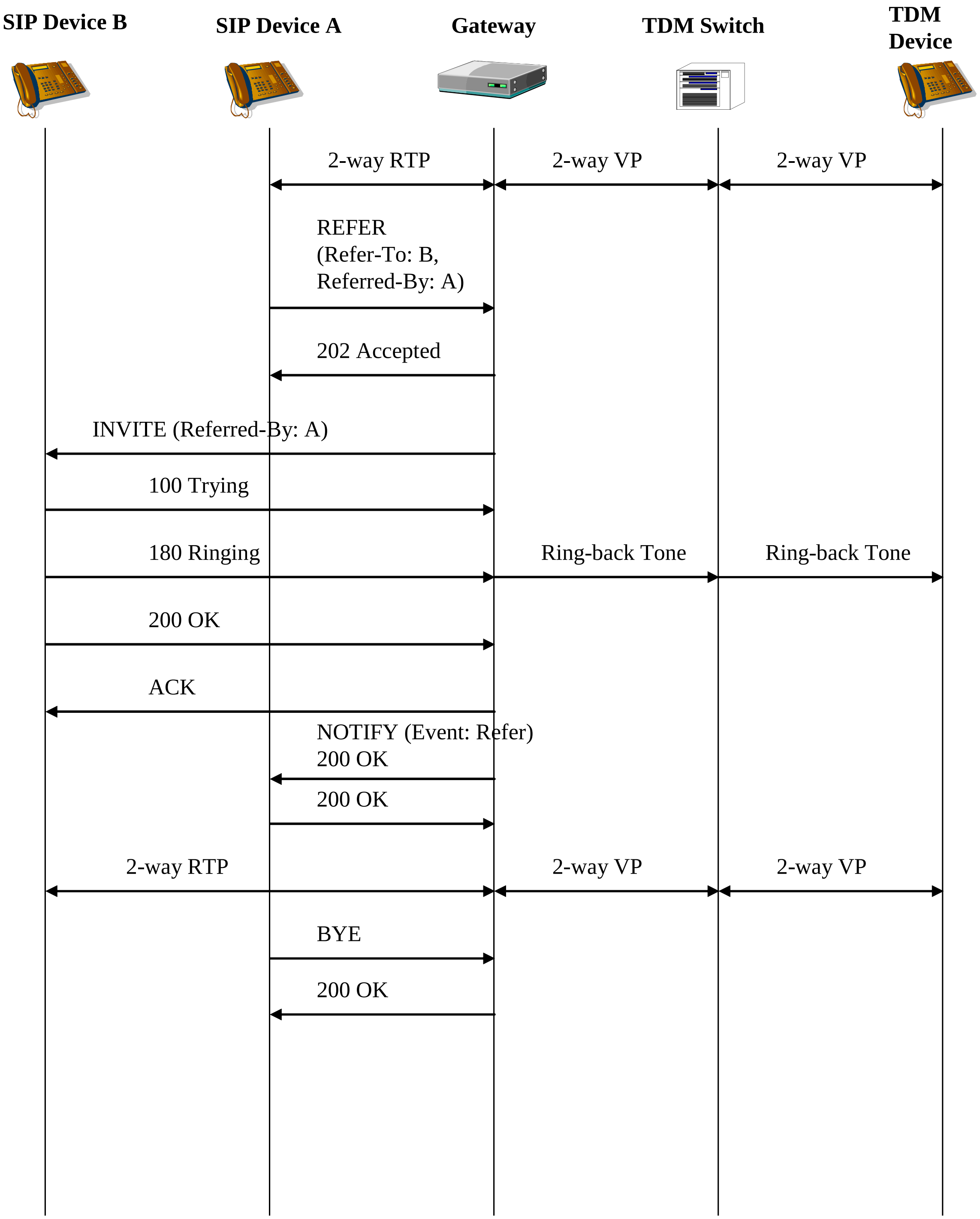
TDM Call Drop

If a ‘disconnect’ state is detected on the TDM network call (call terminates, dial-tone detected, disconnect-event), then the TDM call is terminated and a BYE request is sent.



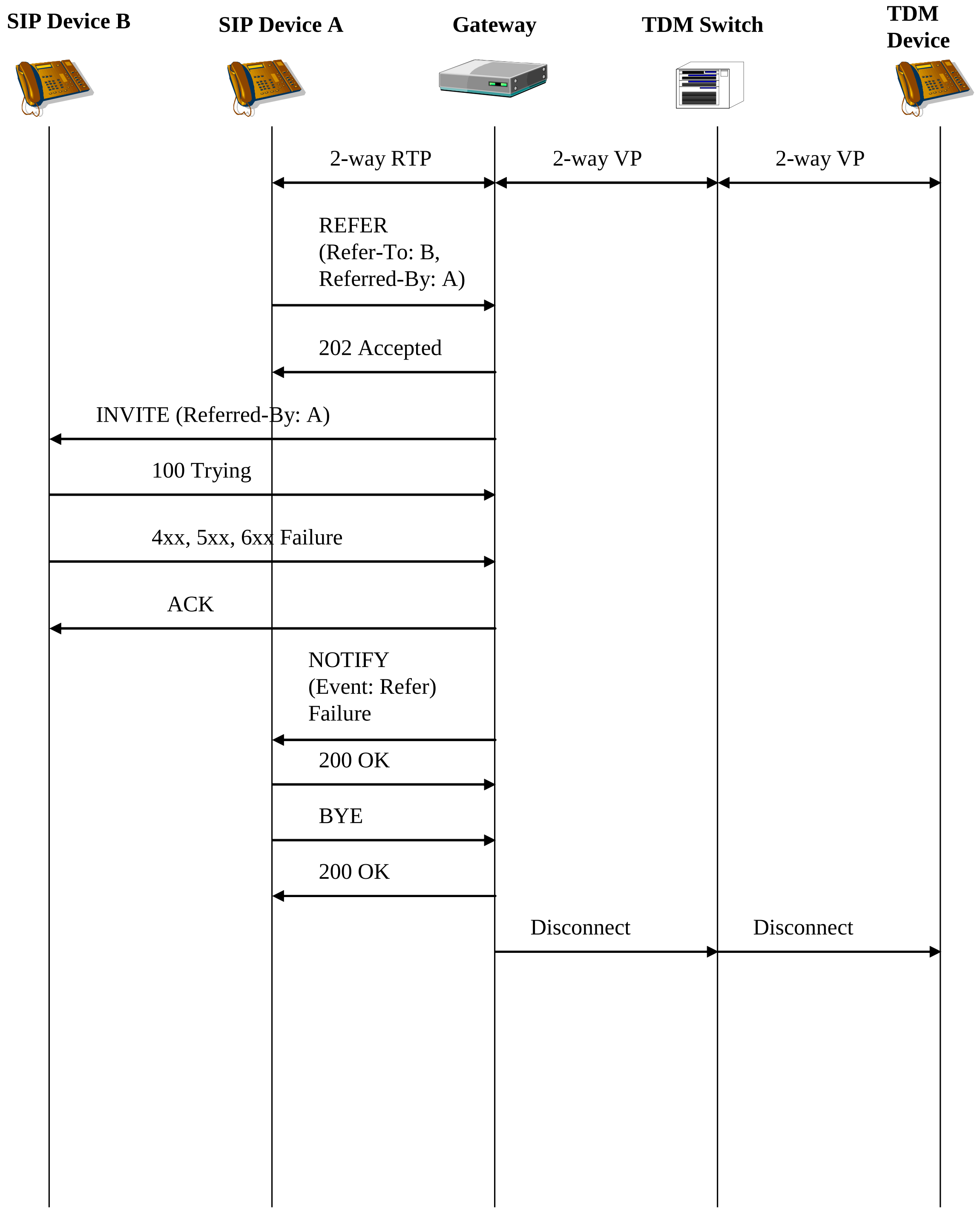
Unsupervised Transfer Success - VOIP Target - REFER

A VOIP device uses the REFER method to request that the gateway terminate the initial VOIP call and replace it with a new VOIP call. Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request.



Unsupervised Transfer Failure - VOIP Target - REFER

If the VOIP destination responds with an error, then the TDM call is dropped. Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request.

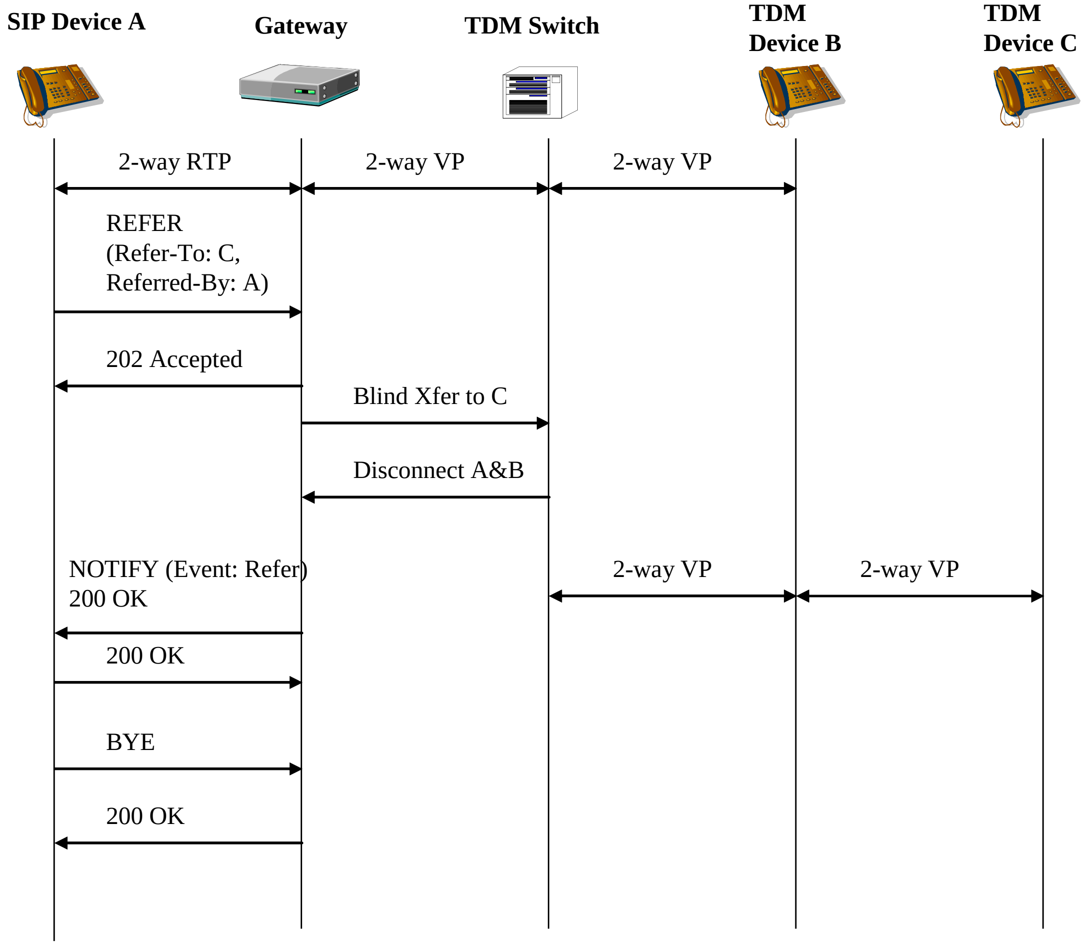


Unsupervised Transfer – TDM Target

This section contains the call scenarios in which a VoIP device is requesting a supervised transfer of one device on the TDM network to another device on the TDM network. In other words, the target of the transfer request from the VoIP network is on the TDM network.

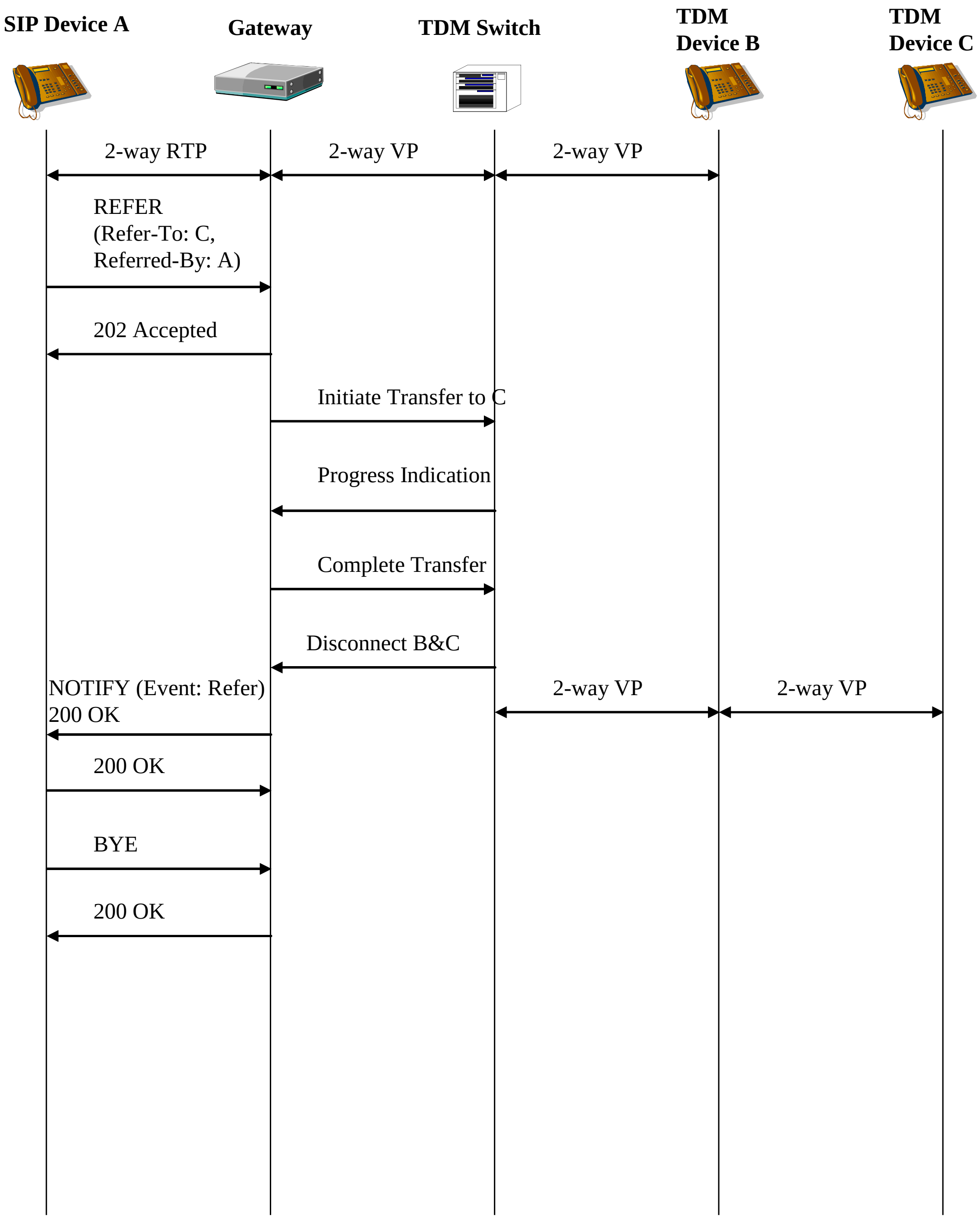
Unsupervised Transfer Success - TDM Target - REFER

If the transfer target (in this case, C) is a destination on the gateway (destination host is the gateway), then the transfer takes place on the TDM network, not on the VOIP network. In this case, the original TDM network party B is transferred to the specified TDM network party C. Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request.



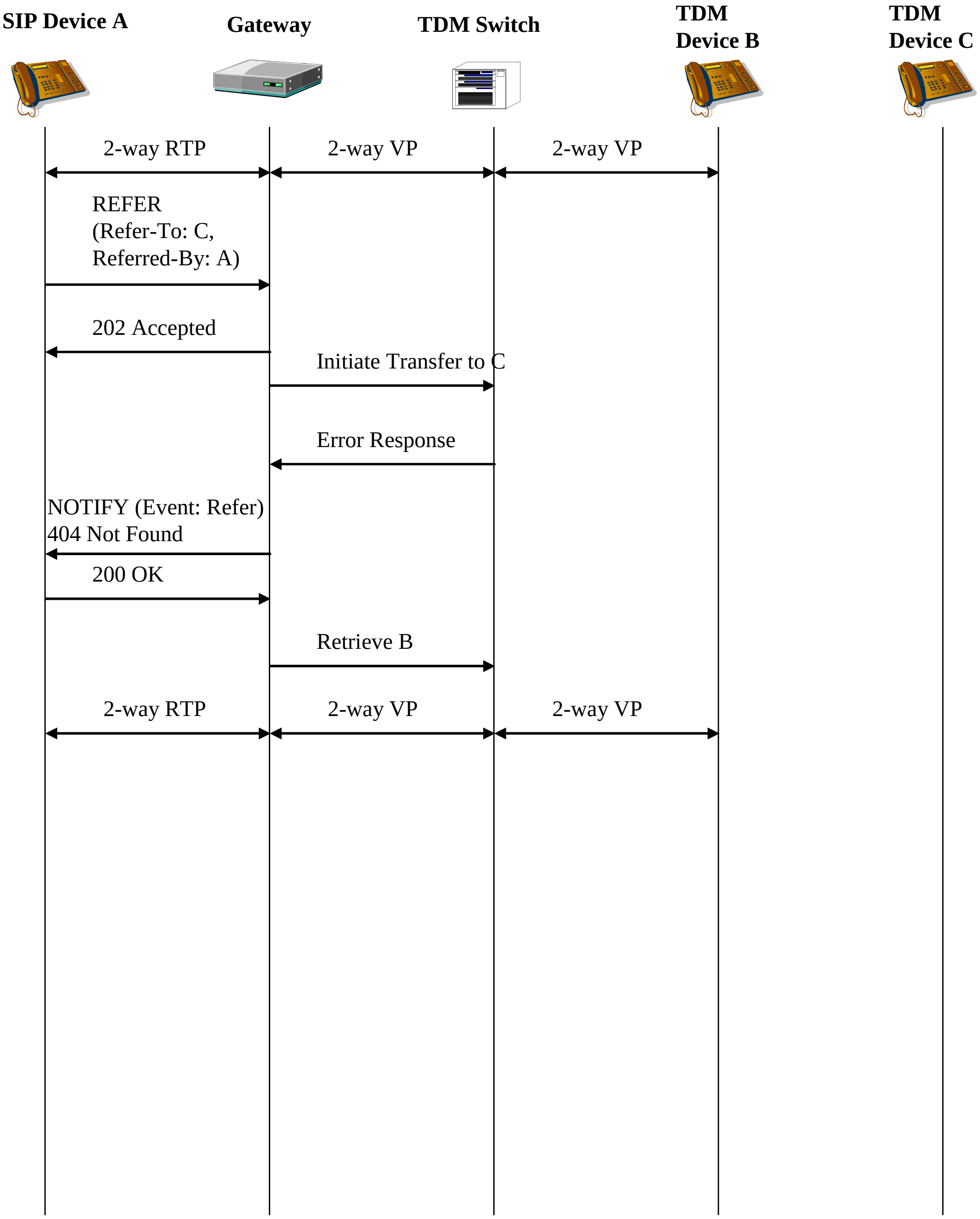
Unsupervised Transfer Success - TDM Target - REFER

The gateway waits for a successful progress indication from TDM device C before completing the transfer. A successful progress indication is a timeout (no tones detected), voice activity, ring-back tone or an ISDN progress indication. If a successful progress indication is received, the transfer is completed. (TDM switches will not transfer calls to busy extensions). Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request.



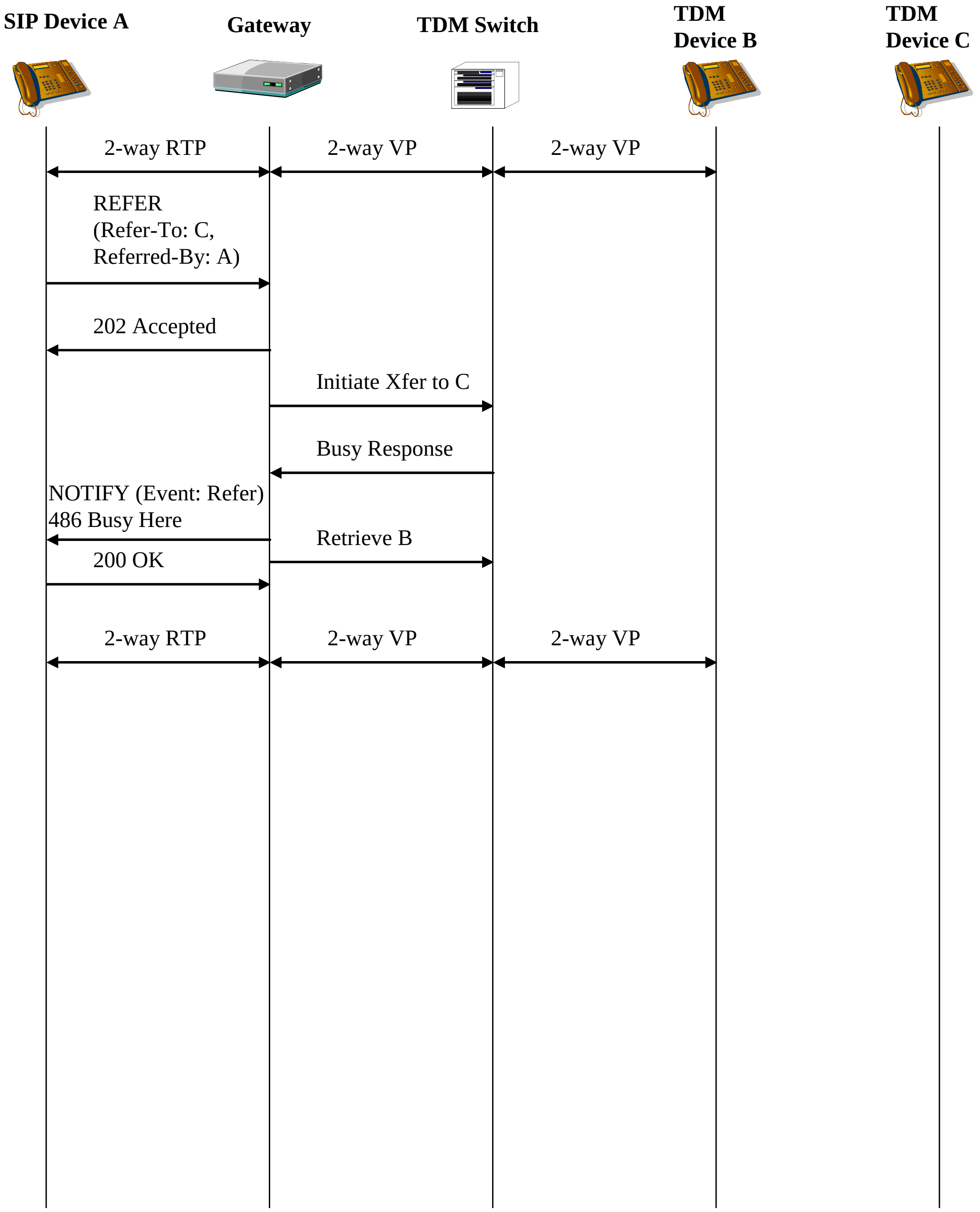
Unsupervised Transfer Failure - TDM Target – REFER Error Response

The gateway waits for a successful progress indication. If an error-response is received from the TDM network, then the original TDM call is retrieved and the transfer fails. Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request. If this occurs, then the gateway will retrieve TDM Device B and then terminate the call.



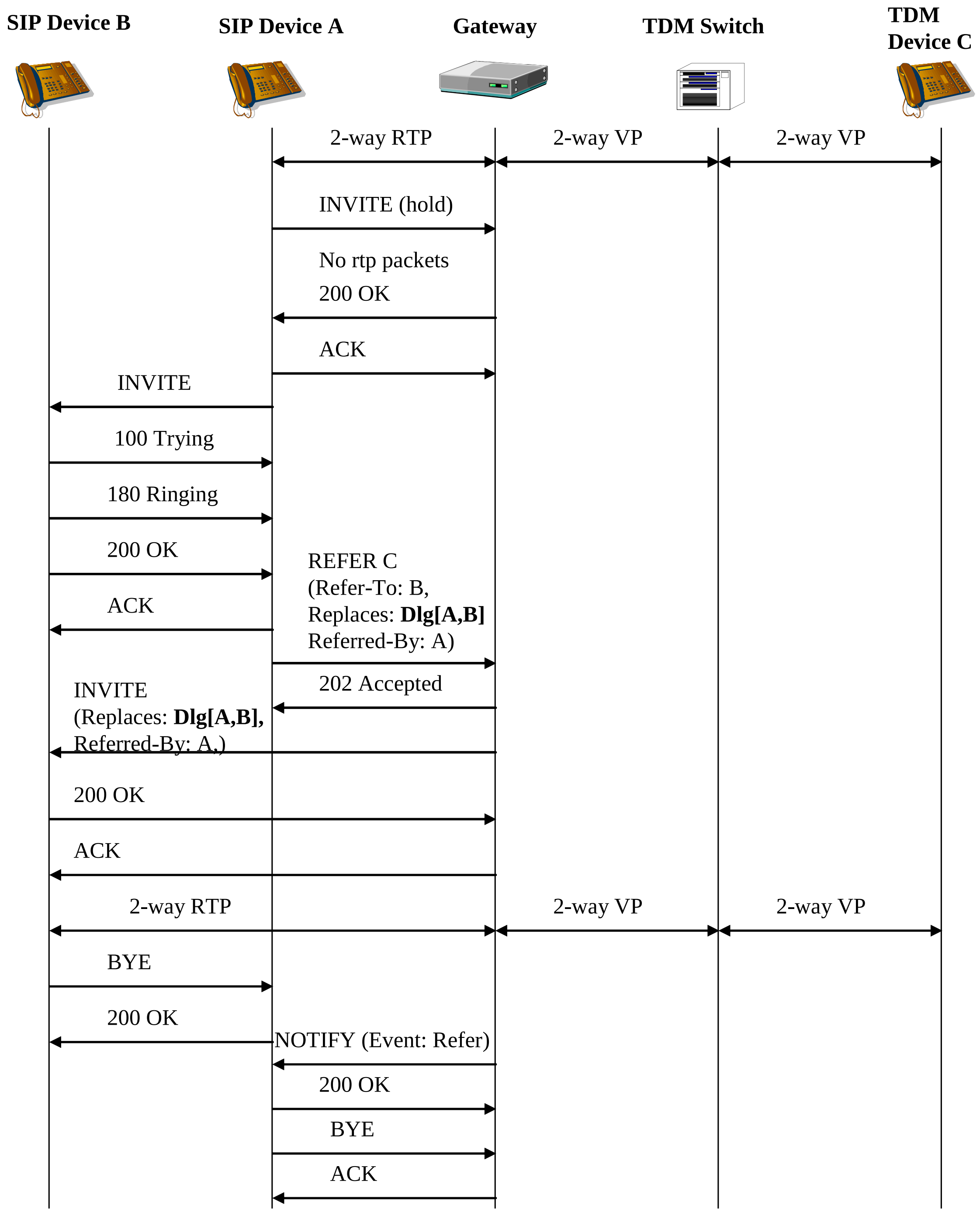
Unsupervised Transfer Failure - TDM Target – REFER Busy Response

The gateway waits for a successful progress indication. If a busy-response is received, then the original TDM call is retrieved and the transfer fails. A busy-response may be busy-tone, an ISDN busy-disconnect, or an ISDN busy-reroute request. Note: SIP Device A may send the BYE immediately after it receives the 202 Accepted for the REFER request. If this occurs, then the gateway will retrieve TDM Device B and then terminate the call.



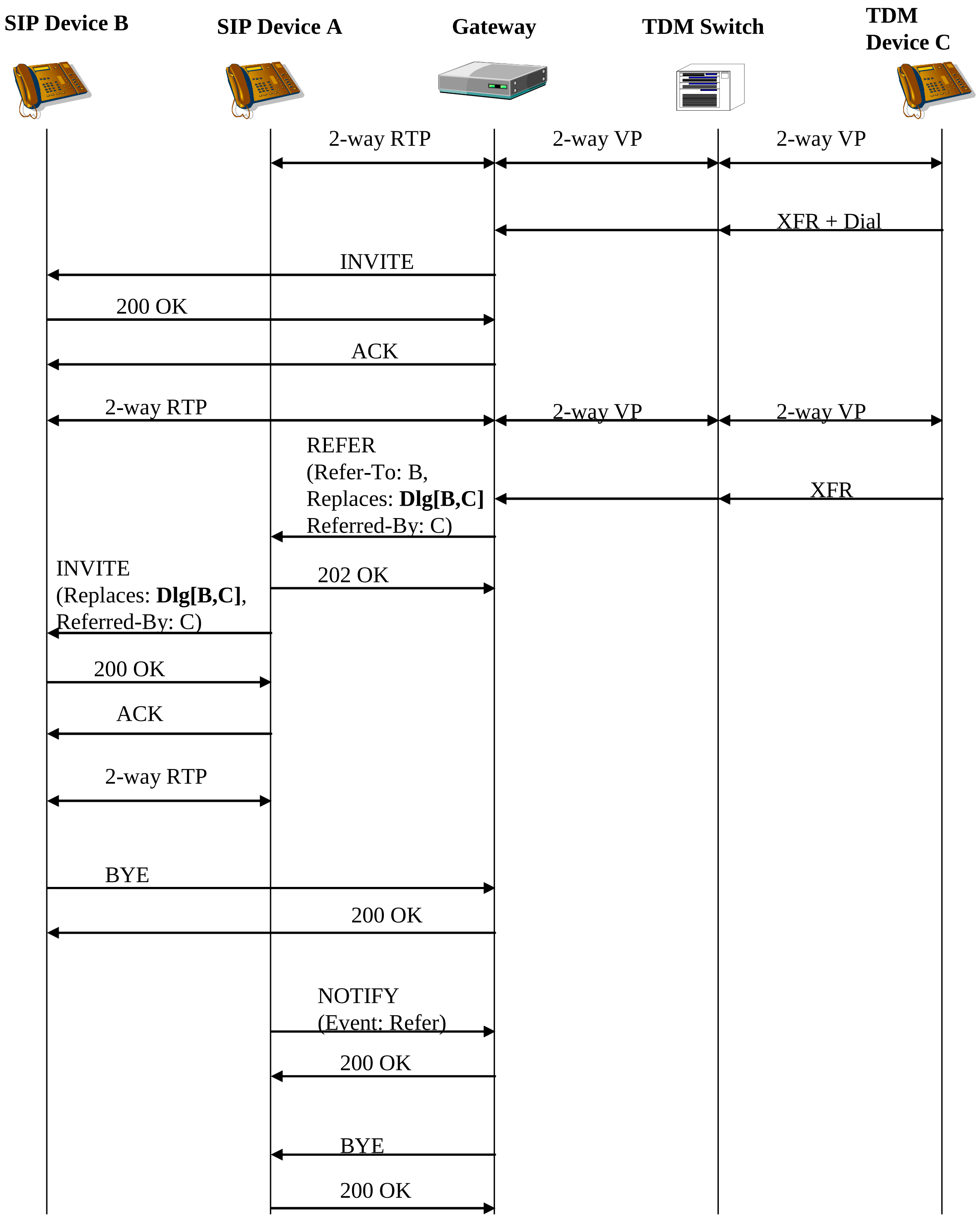
Supervised Transfer Success - VOIP Target, Gateway is Transferee

A consultation transfer to a VOIP target replaces on VOIP endpoint with another on the TDM call. The gateway serves as the Transferee.



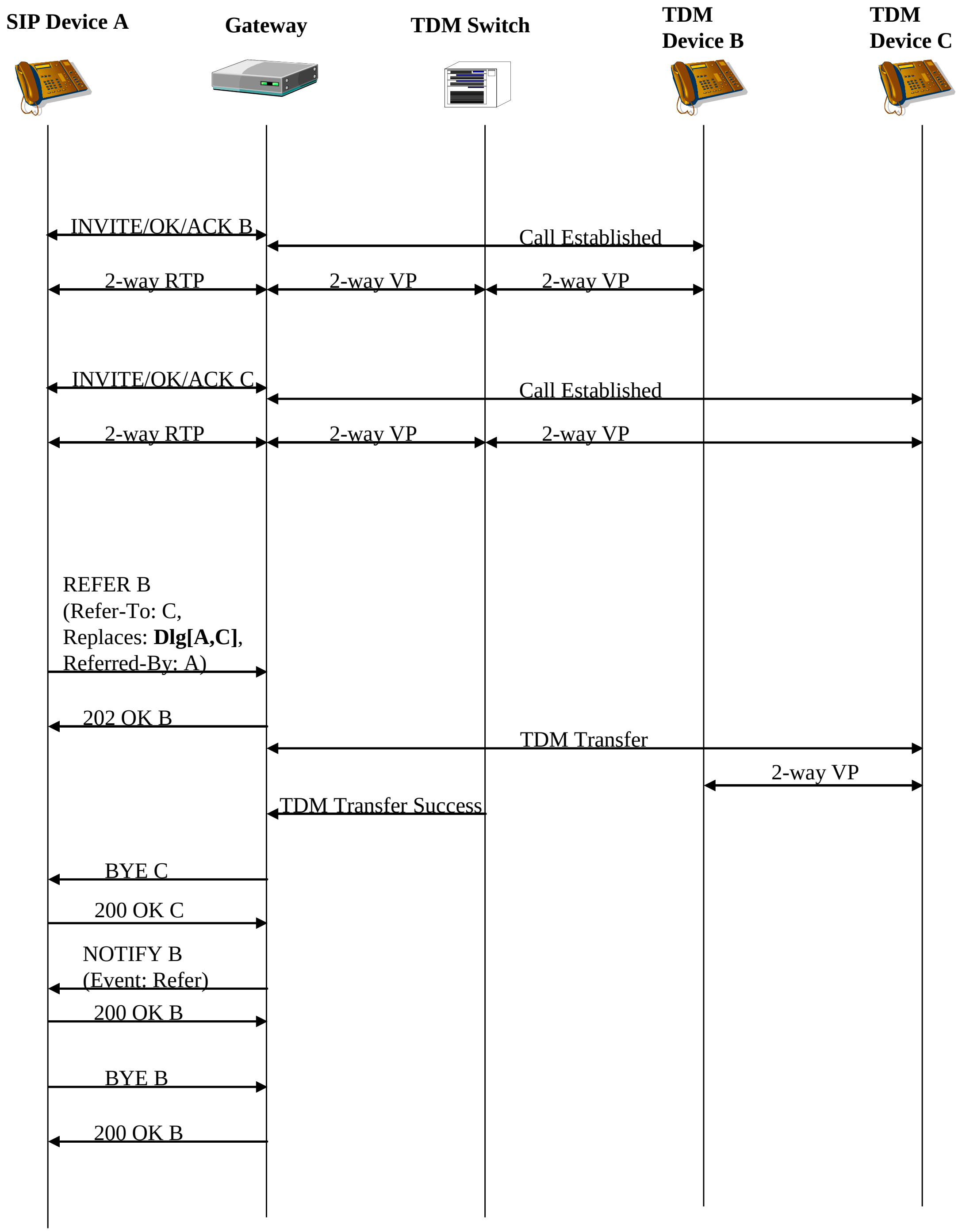
Supervised Transfer Success - VOIP Target, Gateway is Transferor

The TDM device makes a consultation call to a new VOIP endpoint. The gateway serves as the Transferor.



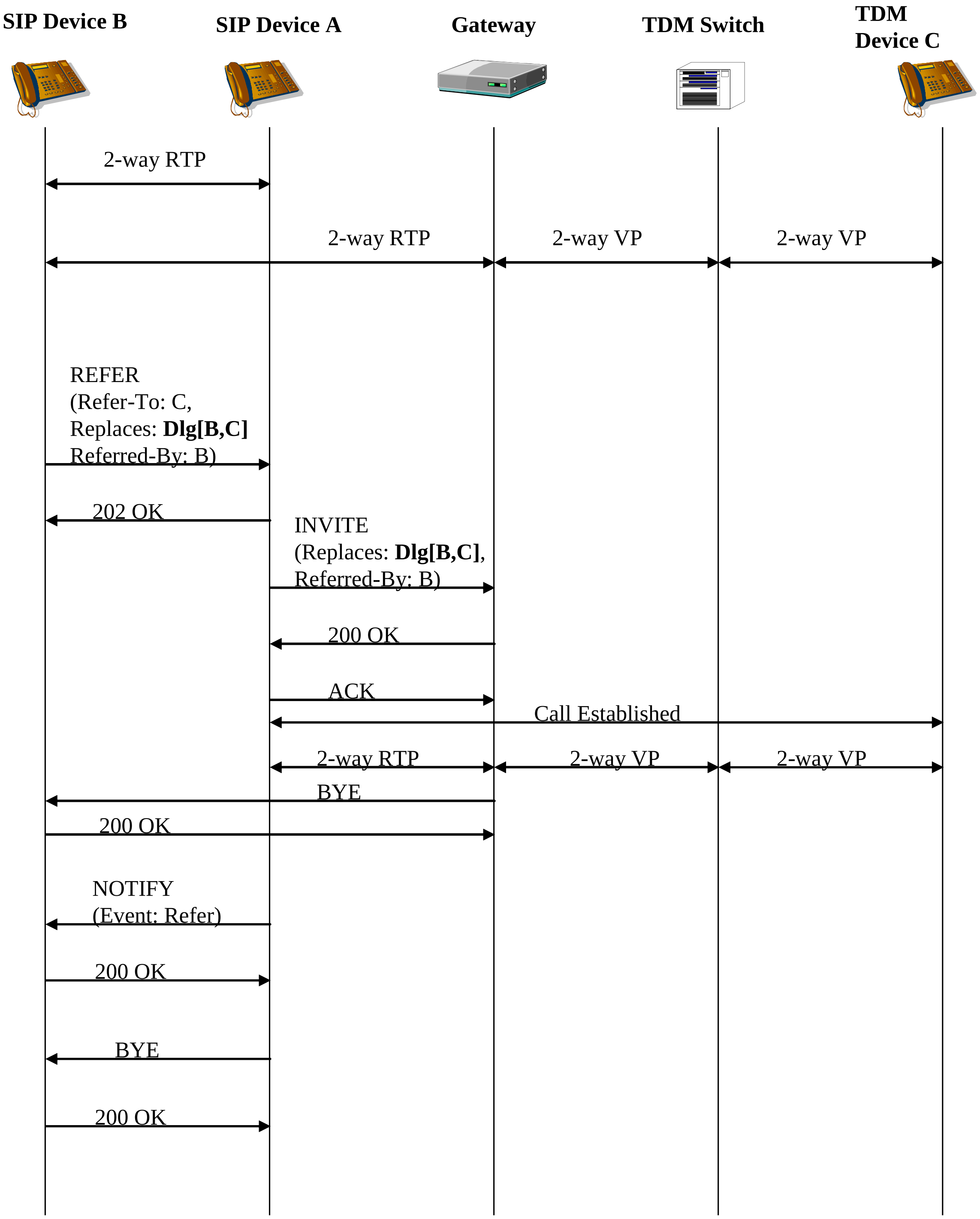
Supervised Transfer Success - TDM Target, Gateway is Transferee and Transferor

SIP Device A makes a consultation call to TDM Device C for the purpose of transferring TDM device B to TDM device C. The Gateway severs as both the Transferee and the Transfer target.



Supervised Transfer Success - TDM Target, Gateway is Target

SIP Device B make a consultation call to TDM device C for the purpose of transferring SIP device A to TDM Device C. The Gateway serves as the Transfer Target.



Supervised Transfer – TDM Target, Gateway is Transferee and Target

Supervised transfers of a call to a TDM device must be supported in a different manner than unsupervised transfers. Unsupervised transfers are handled completely by the Media Gateway. The party requesting the unsupervised transfer does not have access to the audio stream of the transfer target, and therefore cannot interact with the transfer target. The Media Gateway initiates and completes the unsupervised transfer when a REFER request is received.

Supervised transfers require that the party requesting the transfer have access to the audio stream of the transfer target, so that the requesting party can interact with the transfer target before completion of the transfer. This requires a two-step transfer process, of which SIP currently does not support. SIP does support the use of a Re-INVITE to place a call on hold so that the transfer target of a supervised transfer can be called, but this does not provide the Media Gateway with enough information to know that the hold request and the subsequent INVITE of the transfer target is indeed a request to initiate a transfer, and not just a request to place the first call on hold and originate another call to the transfer target.

Because of this, a proprietary header is introduced. The header is called 'Transfer-From' and uses the following format:

```
Transfer-From  = "Transfer-From" HCOLON callid *(SEMI xfr-from-param)
xfr-from-param = to-tag / from-tag / generic-param
to-tag         = "to-tag" EQUAL token
from-tag       = "from-tag" EQUAL token
```

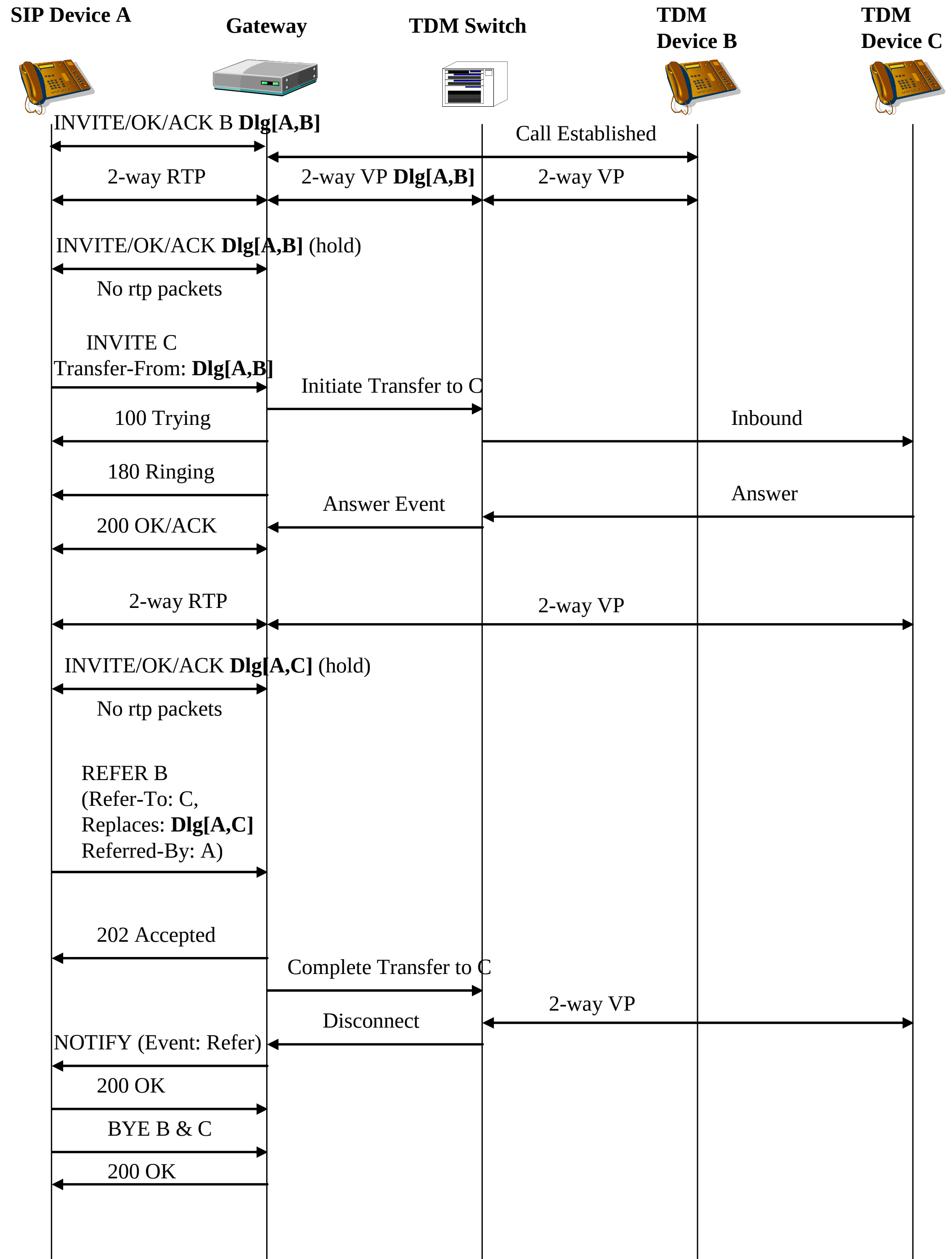
The 'Transfer-From' header follows the same format and processing as a 'Replaces' header, meaning that the header is processed by the receiver as if the tags had been received in a received request. Therefore, the to-tag must contain the local (receivers) tag and the from-tag must contain the remote (senders) tag.

The 'Transfer-From' header will be valid only in a new dialog INVITE to the Media Gateway. This header will specify the dialog that is to be used as the transferee in the supervised transfer. If the Media Gateway can match the dialog specified by the 'Transfer-From' header, then the corresponding TDM call will be used as the transferee party of the supervised transfer. The Media Gateway will initiate the transfer on the transferee call and originate a call to the transfer target specified by the INVITE.

The requesting party can then abort the transfer, by sending a BYE to the transfer target dialog, or it can complete the transfer, by sending a REFER request to the transferee dialog (requesting a transfer to the transfer target).

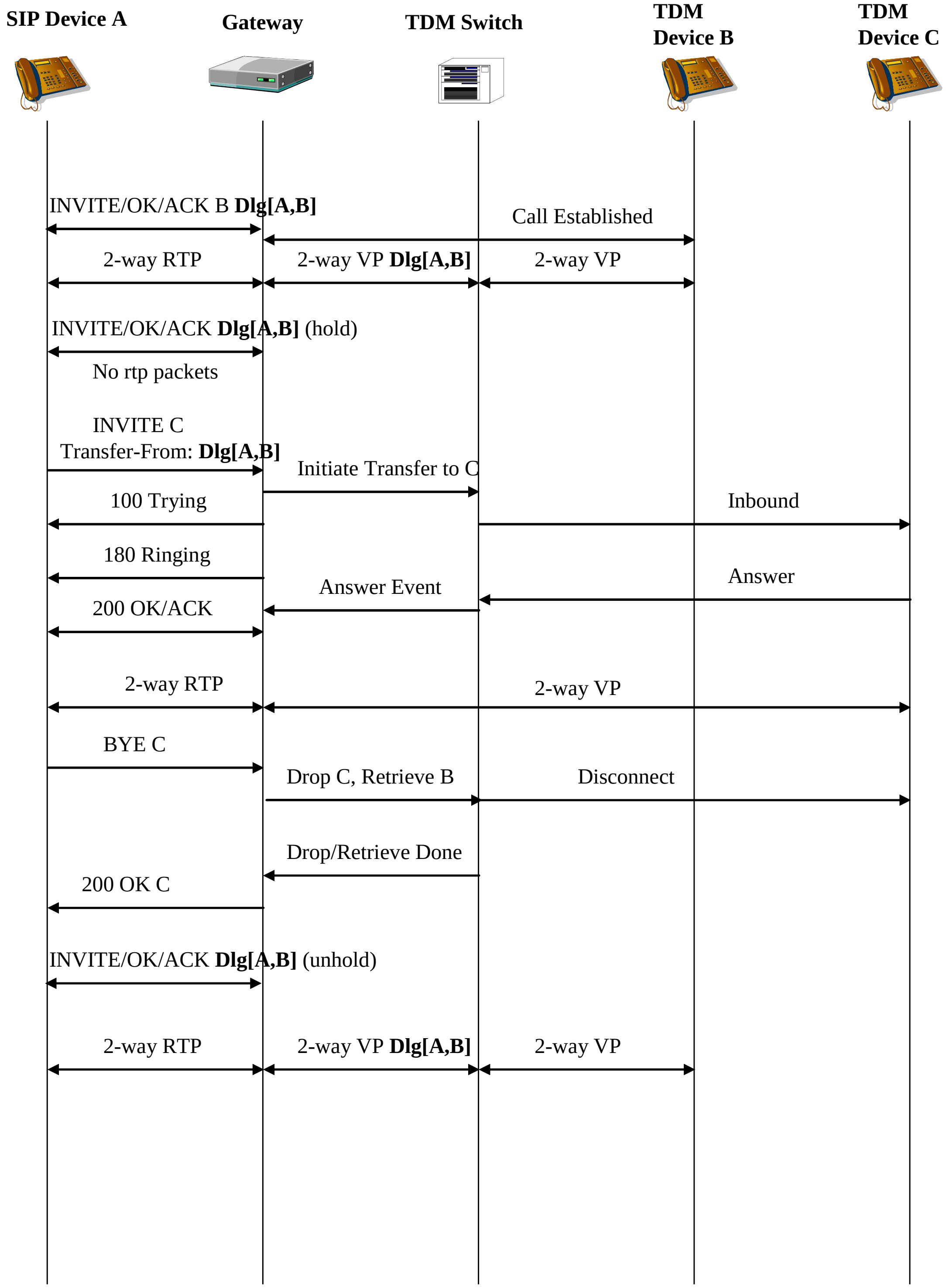
Supervised Transfer Success - TDM Target, Gateway is Transferee and Target

In the following diagram, SIP Device A is the transferor (transfer requestor), TDM Device B is the transferee, and TDM Device C is the transfer target. SIP Device A first establishes a call to TDM Device B (Dlg[A,B]). SIP Device A then places Dlg[A,B] media on-hold using a Re-INVITE. SIP Device A then initiates the transfer by sending an INVITE to TDM Device C, the transfer target, specifying Dlg[A,B] as the 'Transfer-From' dialog. Media Gateway matches the 'Transfer-From' dialog to the call with TDM Device B, and therefore initiates a transfer to TDM Device C from the call with TDM Device B. SIP Device A can then interact with the transfer target (TDM Device C) before completing the transfer with a REFER request to Dlg[A,B] to TDM Device C.



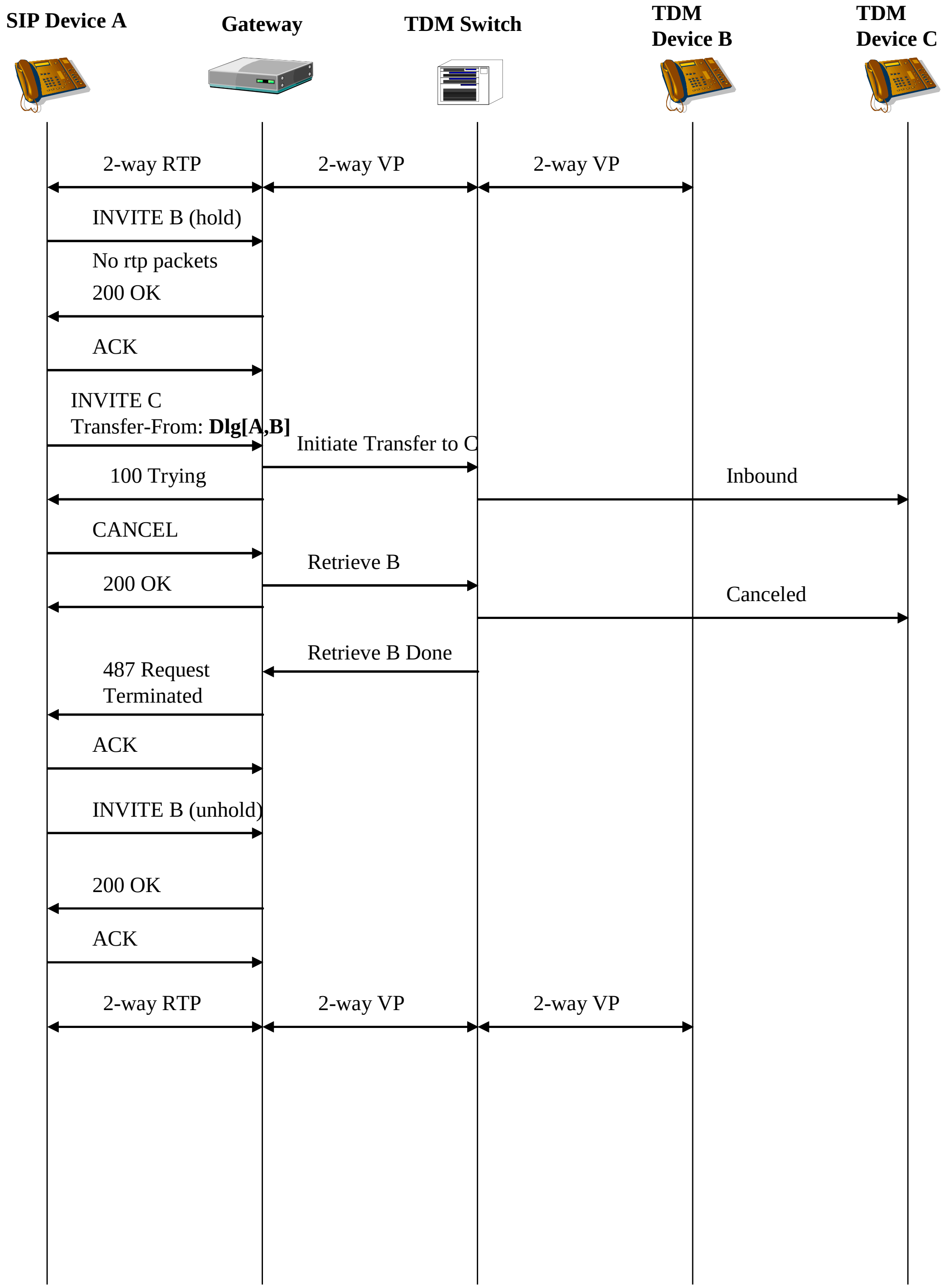
Supervised Transfer Failure - TDM Target – Canceled

The following diagram shows SIP Device A aborting the supervised transfer to TDM Device C. The original call between SIP Device A and TDM Device B is re-established.



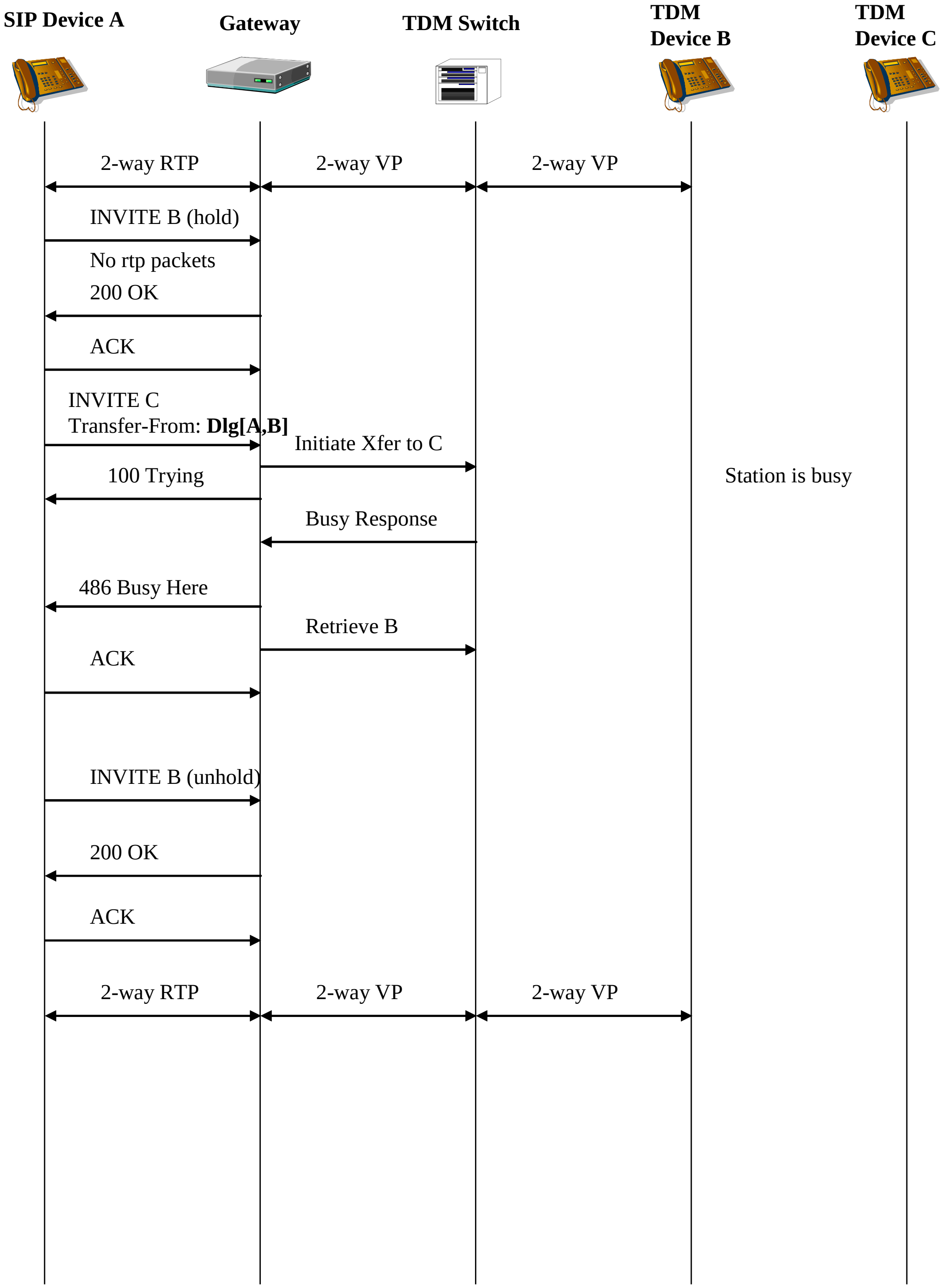
Supervised Transfer Failure - TDM Target – Canceled

If the VOIP device cancels the transfer, which could be a result of a ring-no-answer timeout, then the gateway retrieves the original call.



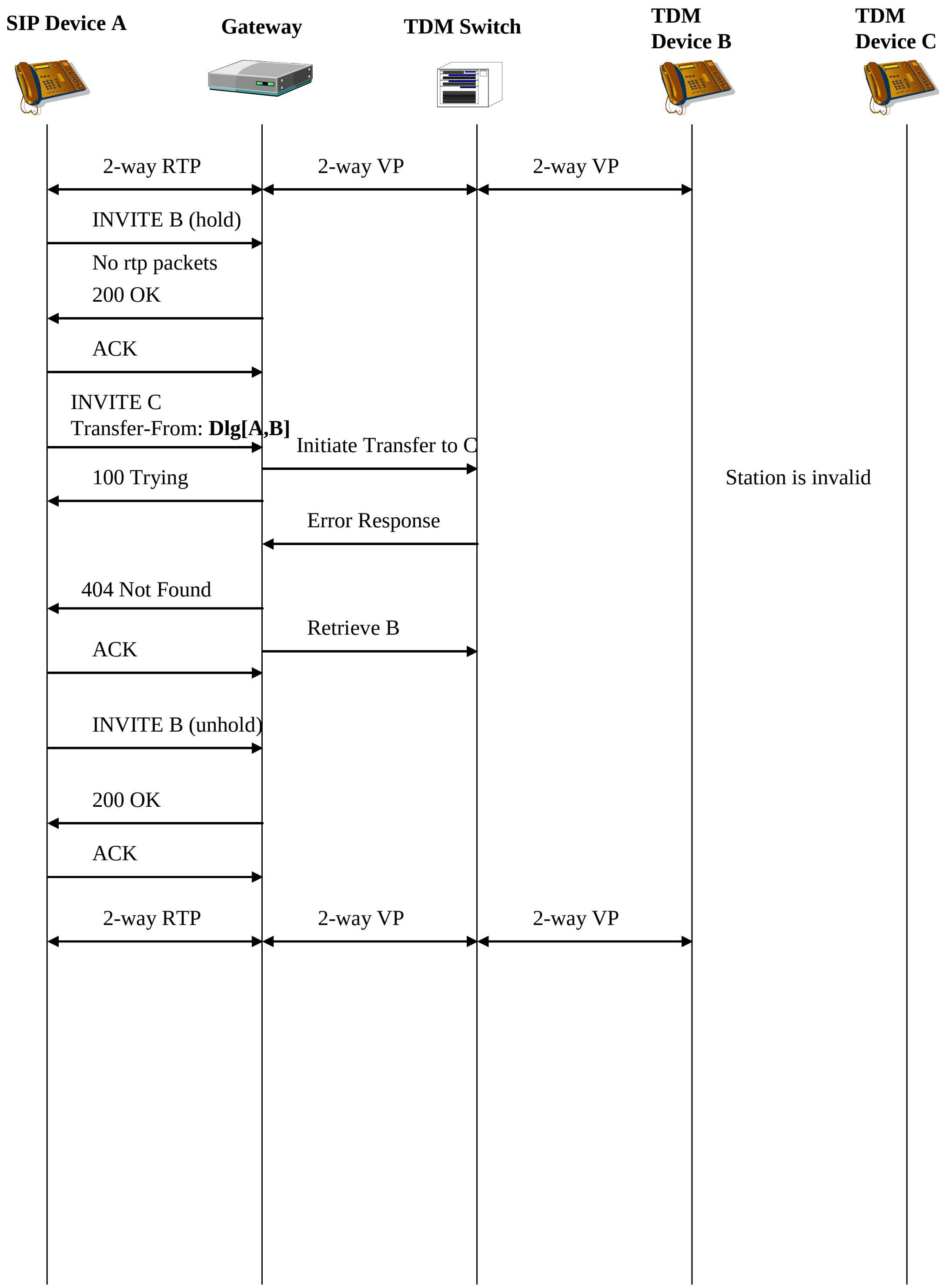
Supervised Transfer Failure - TDM Target – Busy Response

If a busy-response is received from the transfer-target, then the transfer fails and the original call is retrieved.



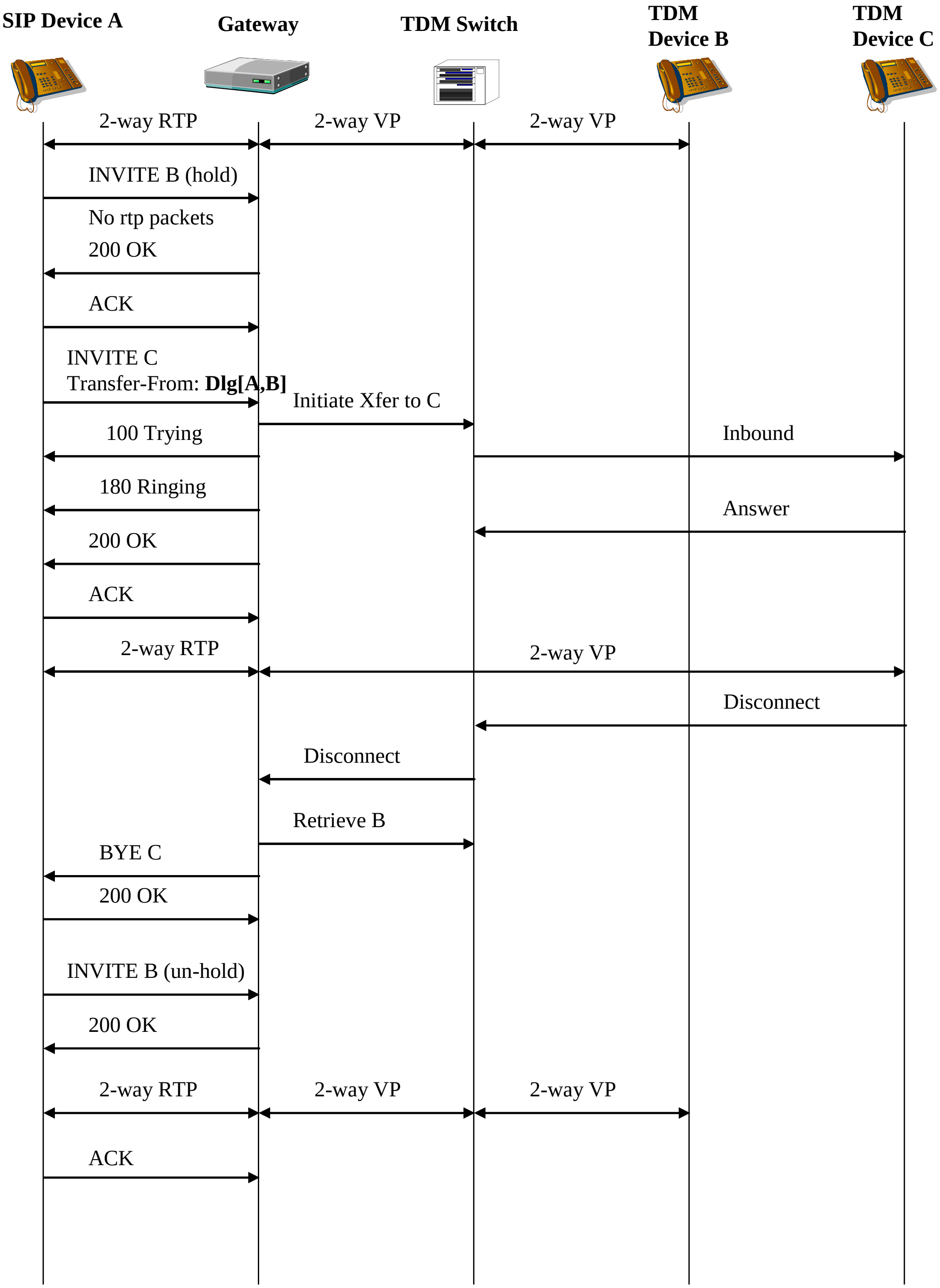
Supervised Transfer Failure - TDM Target – Error Response

If an error response is detected on the transfer, then the transfer fails and the original call is retrieved.



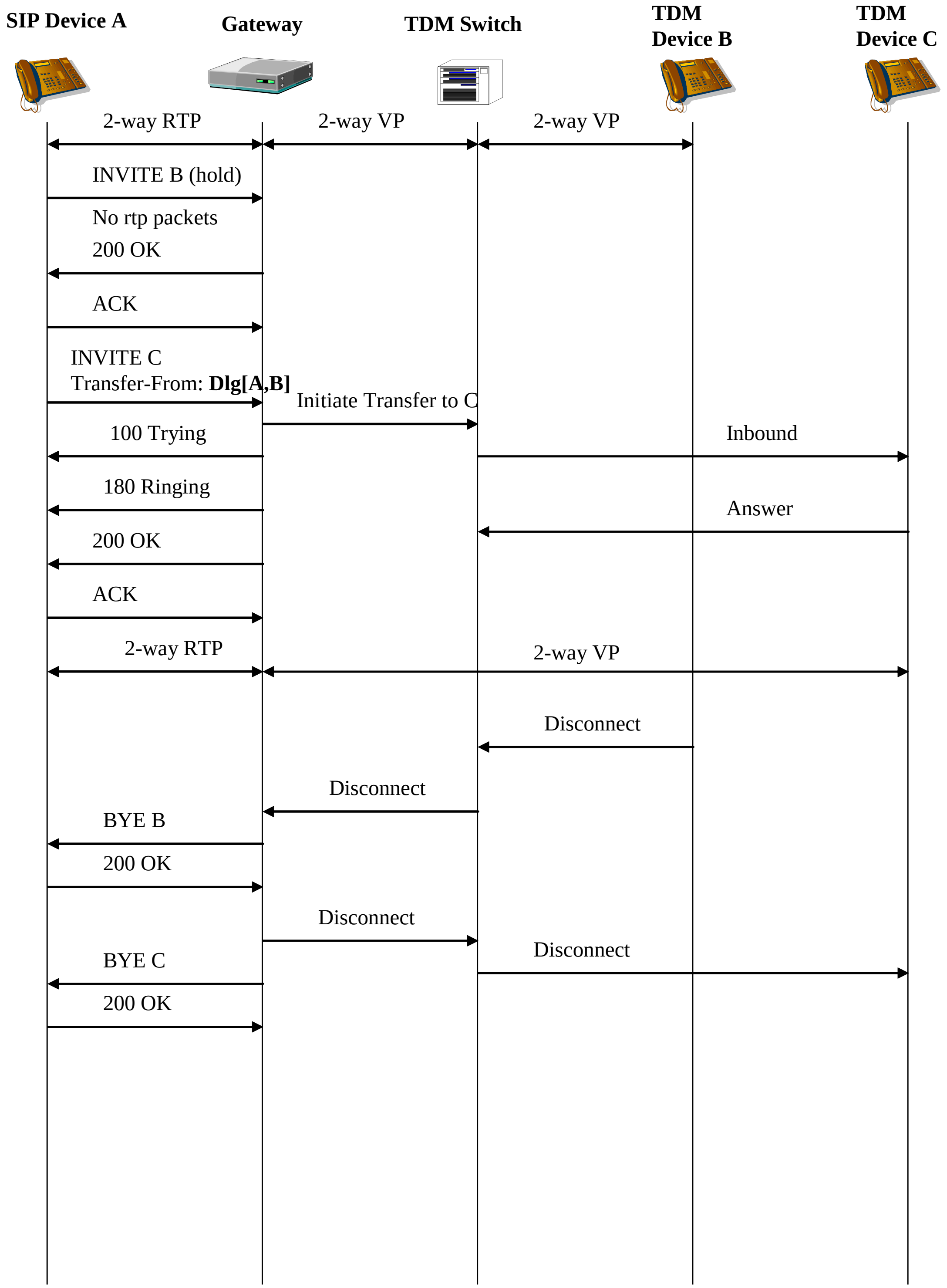
Supervised Transfer Failure - TDM Target Drops

If the target of the supervised transfer drops, then the gateway automatically re-establishes the call with the original TDM network party.



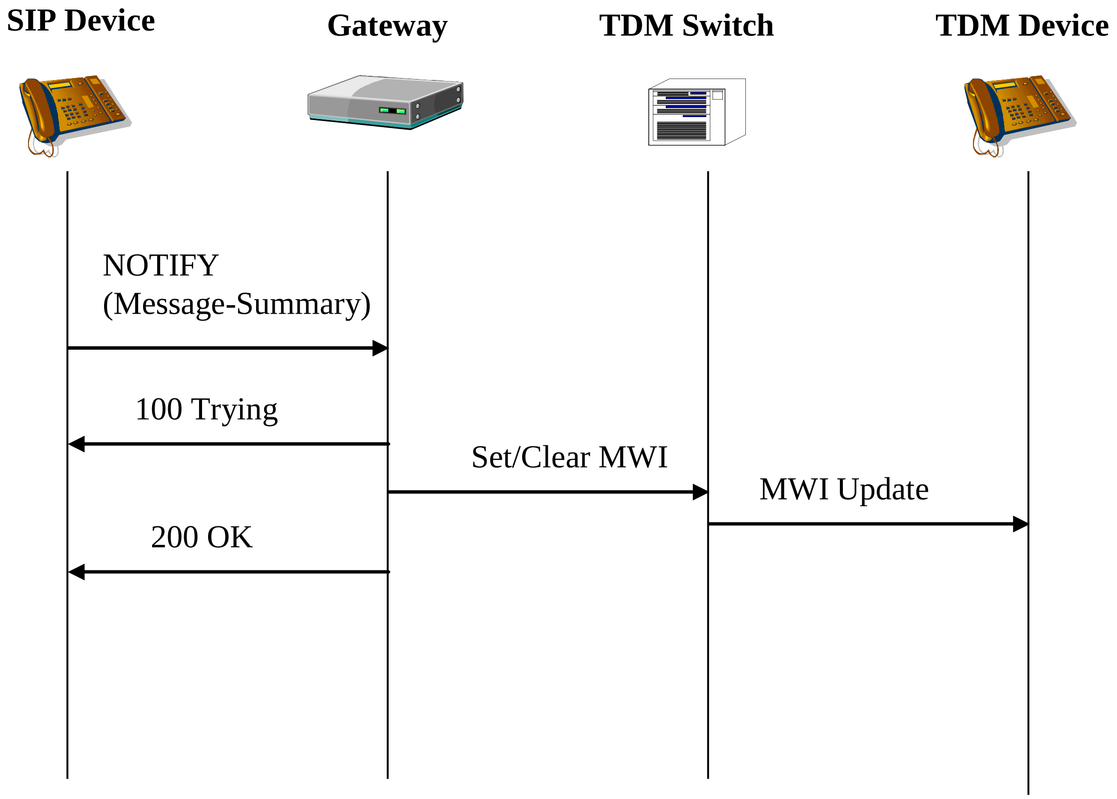
Supervised Transfer Failure - TDM Transferee Drops

If the transferee of the supervised transfer drops, then the gateway drops both the transferee and the transfer-target calls.



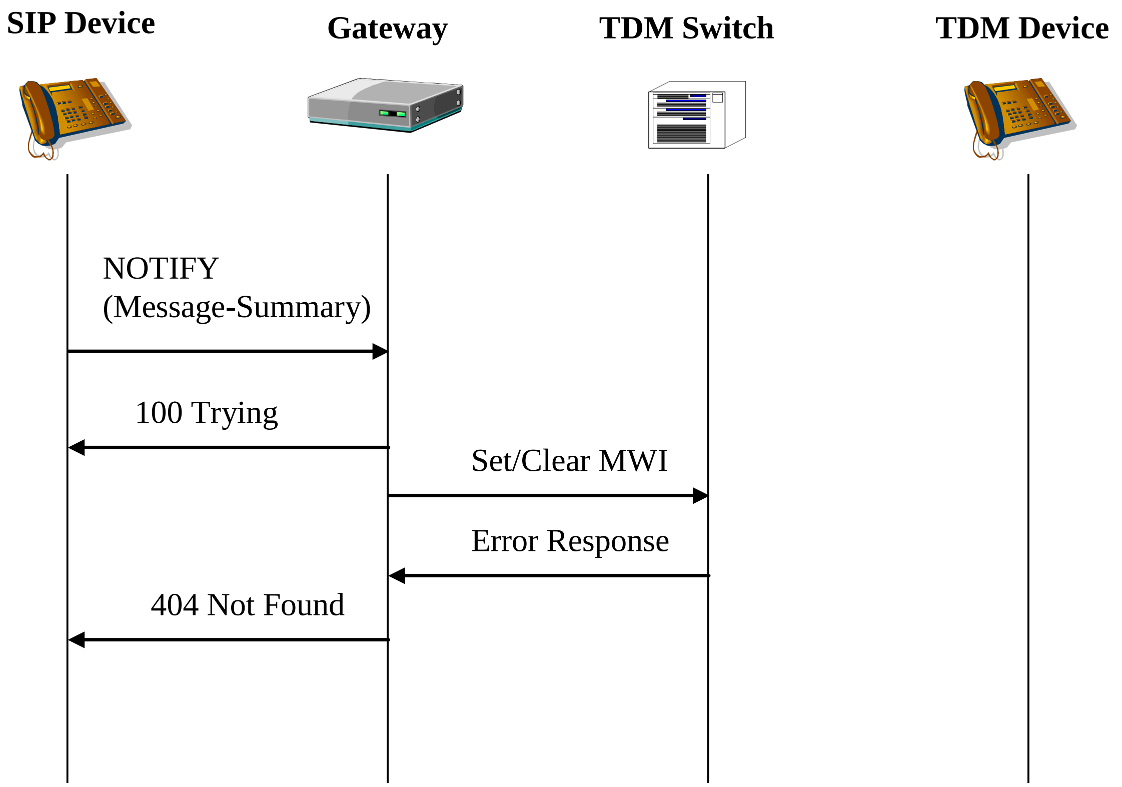
MWI - VOIP to TDM Success

A VOIP NOTIFY request may be made for Message-Waiting to request an MWI set/clear of a TDM network device. The NOTIFY request must specify the destination TDM network device in the *To* header (see section 0).



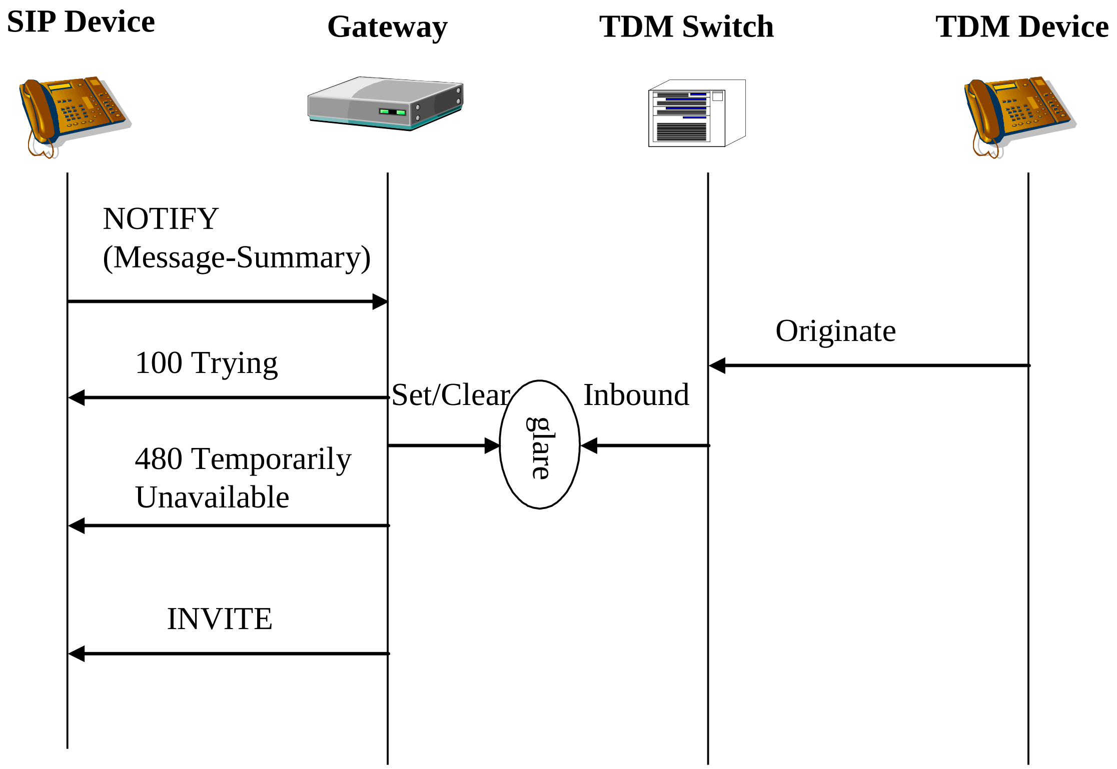
MWI - VOIP to TDM Failure

If the MWI set/clear fails, an error response is sent for the NOTIFY request.



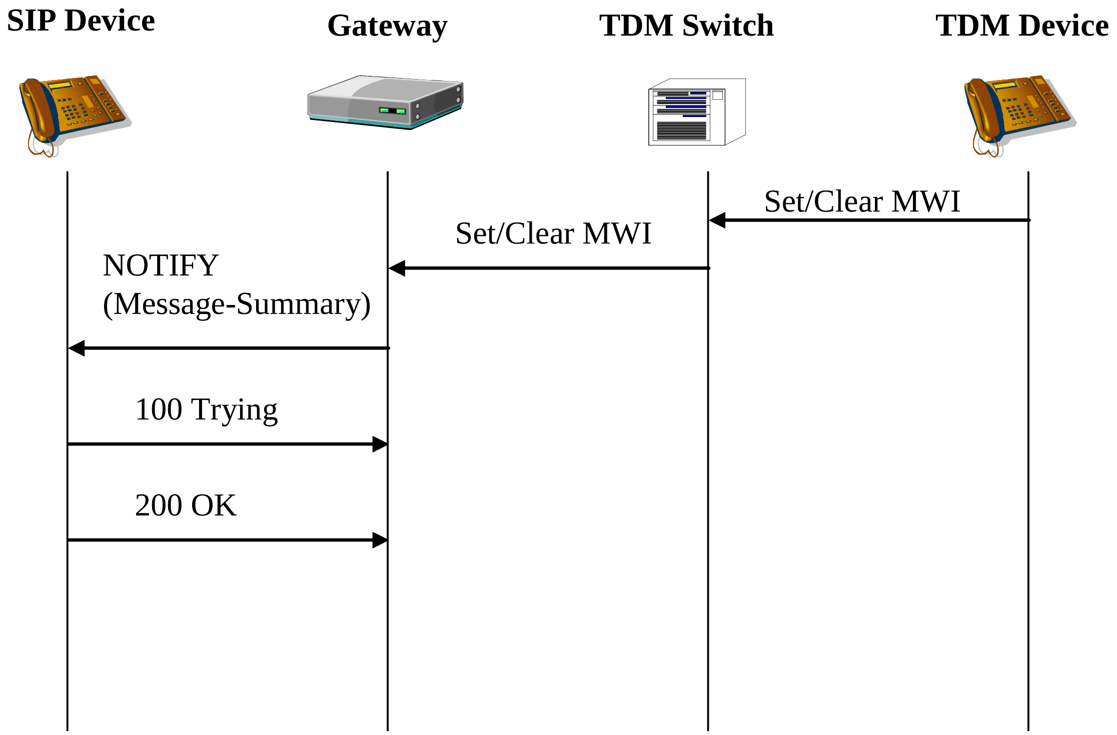
MWI - VOIP to TDM Failure - Glare

If the MWI set/clear fails because of a glare condition, an error response is sent for the NOTIFY request, and a new INVITE request is generated for the new call.



MWI – TDM to VOIP Success

A VOIP NOTIFY request may be made for Message-Waiting to request an MWI set/clear of a VOIP network device. The NOTIFY request must specify the destination VOIP network device in the *To* header (see section 0).



T.38 Fax - TDM to VOIP

A TDM to VOIP T.38 Fax call is initiated by a fax machine calling a telephony port on the gateway. The SDP information placed in the INVITE request to the VoIP network includes the fax media type and control information. The following are set in the SDP for every call to enable T.38 Fax operation later in the call if a fax machine is detected:

```
m=image 0 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

The call flow proceeds in the same manner as a voice call and the RTP stream for voice is started. When the gateway detects CNG tone from the fax machine, it passes the tone through the voice stream and also generates RFC2833 messages for the tone if digit relay is enabled. The CNG tone can be used by the VOIP destination to distinguish between voice and fax calls and to re-route fax calls in whatever manner is appropriate for the application. The switchover to T.38 relay is initiated by the VOIP destination by issuing an INVITE request to the gateway which contains SDP information that contains a non-zero port number for the T.38 Fax media type and a zero port number for all other media types. The gateway will send a 200 OK message with SDP information that contains the port number that it will use to transmit T.38 UDPTL. The gateway will then close the RTP voice stream and open the specified RTP/RTCP ports for T.38 UDPTL. The call will proceed in T.38 mode until either endpoint terminates the call with a BYE request.

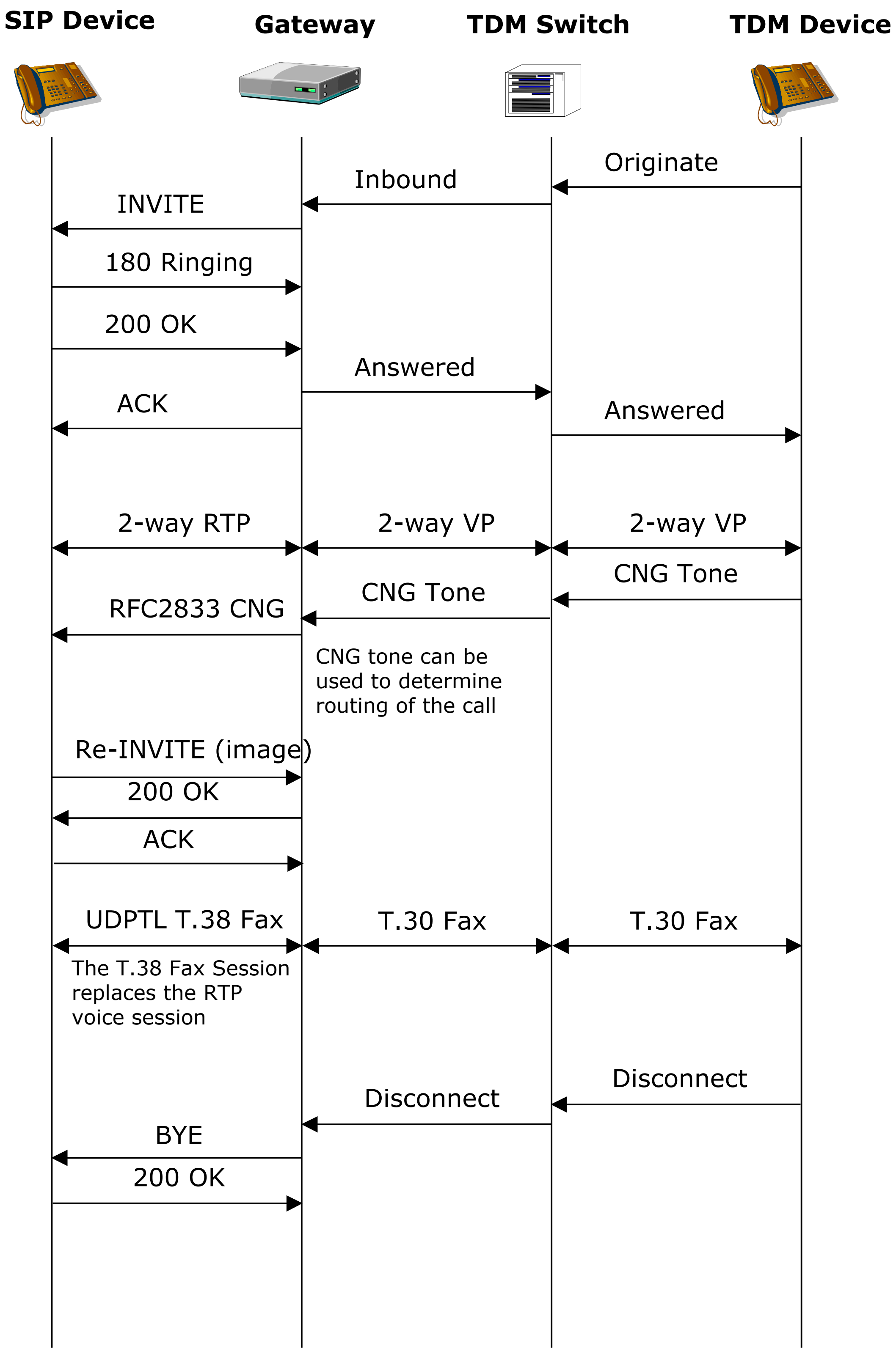
Example SDP information contained in the INVITE request sent by the VOIP destination:

```
m=audio 0 RTP/AVP 0 101 13
m=image 49534 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

Example SDP information contained in the 200 OK message sent by the by the gateway:

```
m=audio 0 RTP/AVP 0 101 13
m=image 49020 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

The gateway does not require that the VOIP destination support T.38 Fax. If T.38 relay is not supported, the VOIP destination will not recognize the CNG tone and not issue the INVITE for fax and the call will proceed in voice mode.


SIP Device
Gateway
TDM Switch
TDM Device
Originate
Inbound
INVITE
180 Ringing
200 OK
Answered
ACK
Answered
2-way RTP
2-way VP
2-way VP
CNG Tone
CNG Tone
RFC2833 CNG
CNG tone can be used to determine routing of the call
Re-INVITE (image)
200 OK
ACK
UDPTL T.38 Fax
T.30 Fax
T.30 Fax
The T.38 Fax Session replaces the RTP voice session
Disconnect
Disconnect
BYE
200 OK

T.38 Fax – VOIP to TDM

A VOIP device that supports T.38 Fax communication may specify the fax media type with a port number of zero in the SDP information contained in the original INVITE request. When this VOIP device places a TDM network call that reaches a fax machine, the call proceeds in the same manner as a voice call and the RTP stream for voice is started. When the fax machine answers the TDM network call, it plays CED tone followed by V.21 flags. The gateway will detect the V.21 flags and send a Re-INVITE request to the VOIP device with SDP information that contains a non-zero port number for the T.38 fax media type and a zero port number for all other media types. Upon reception of the 200 OK response, the gateway will close the RTP voice stream and open the named ports for T.38 UDPTL. The call will proceed in T.38 mode until either endpoint terminates the call with a BYE request.

The VOIP device may also choose to not initially specify its support of a T.38 stream. In this case, the gateway will add the T.38 stream to the SDP that it sends in the INVITE request to switch to T.38.

Example SDP information contained in the initial INVITE request sent by the by the VOIP device, in which VOIP device specifies T.38 stream that is initially disabled:

```
m=audio 23000 RTP/AVP 0 101 13
m=image 0 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

Example SDP information contained in the INVITE request sent by the by the gateway to initiate a T.38 Fax session:

```
m=audio 0 RTP/AVP 0 101 13
m=image 49020 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

Example SDP information contained in the initial INVITE request sent by the by the VOIP device, in which VOIP device does not initially specify a T.38 stream:

```
m=audio 23000 RTP/AVP 0 101 13
```

Example SDP information contained in the INVITE request sent by the by the gateway to switch to a T.38 Fax session:

```
m=audio 0 RTP/AVP 0 101 13
m=image 49020 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

Example SDP information contained in the initial 200 OK message sent by the by the VOIP device:

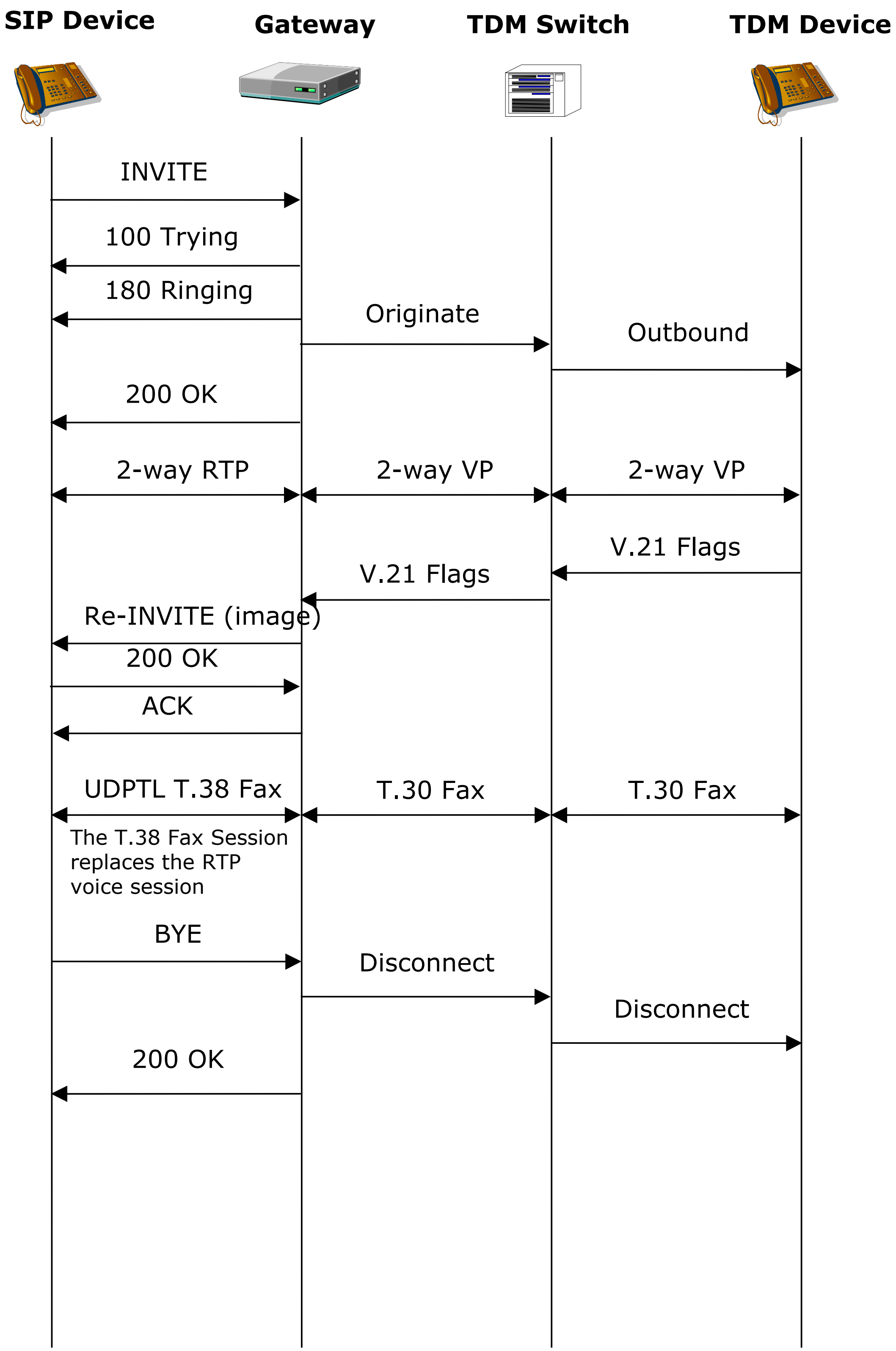
```
m=audio 0 RTP/AVP 0 101 13
m=image 49533 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```

The VOIP device may also request to switch to T.38 by changing its SDP stream from audio to image. For instance, this SDP information could be sent in the original INVITE to the gateway:

```
m=audio 23000 RTP/AVP 0 101 13
```

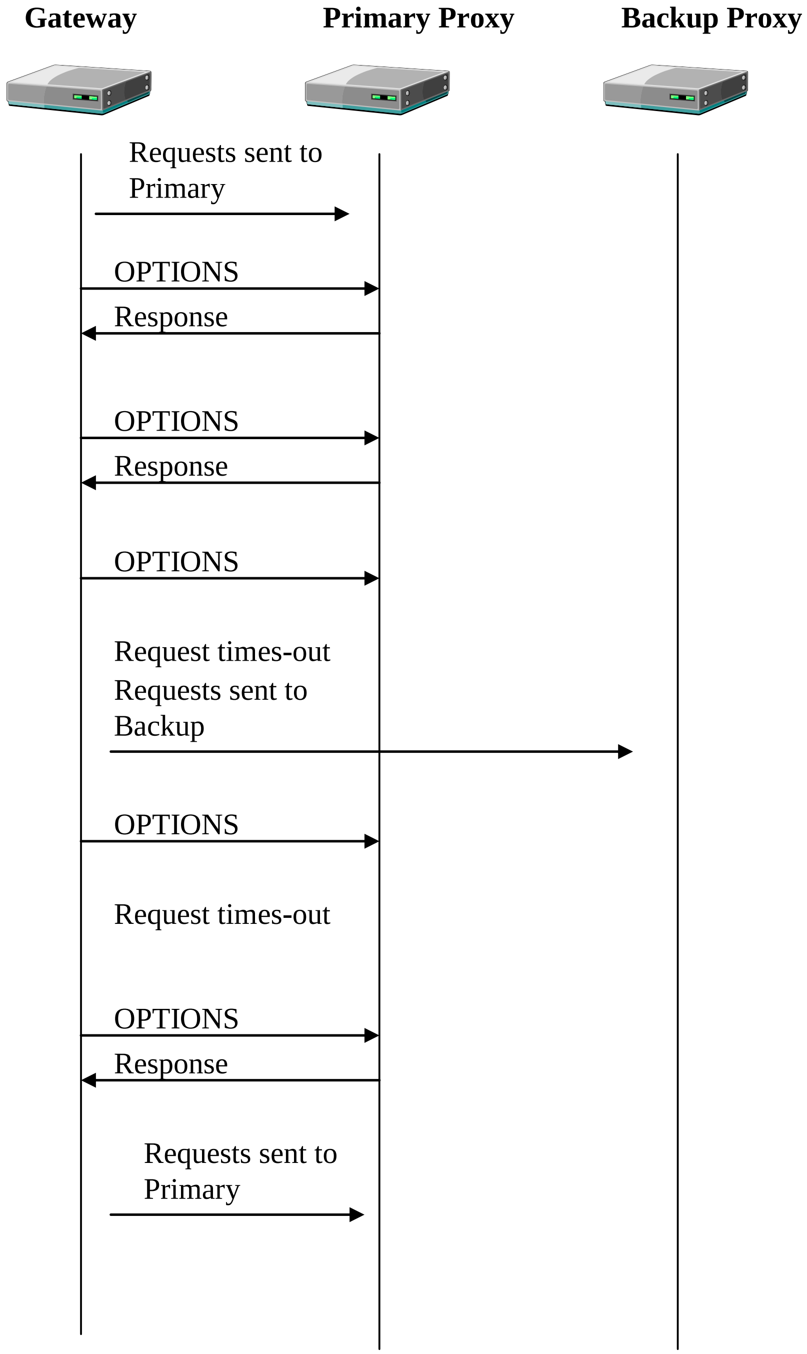
When the VOIP device wishes to switch to T.38, it can change the audio stream to a T.38 stream:

```
m=image 49533 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
```


SIP Device
Gateway
TDM Switch
TDM Device
INVITE
100 Trying
180 Ringing
Originate
Outbound
200 OK
2-way RTP
2-way VP
2-way VP
V.21 Flags
V.21 Flags
Re-INVITE (image)
200 OK
ACK
UDPTL T.38 Fax
T.30 Fax
T.30 Fax
The T.38 Fax Session replaces the RTP voice session
BYE
Disconnect
Disconnect
200 OK

12. Proxy Monitoring

The gateway supports the use of a primary and a backup outbound proxy server. The gateway determines the connection-status of the primary proxy by periodically (period is configurable) sending an OPTIONS request to the primary proxy. If the proxy responds to the OPTIONS request in any manner (error or success), then the gateway continues to use the primary proxy. If the OPTIONS request to the primary proxy times-out with no response, then the gateway switches to using the backup proxy. When using the backup proxy, the gateway continues to send OPTIONS requests to the primary proxy. When the primary proxy responds to the OPTIONS request, the gateway switches back to the primary proxy. Section 0 contains an example of the OPTIONS request which is sent to the primary proxy.


Gateway
Primary Proxy
Backup Proxy
Requests sent to Primary
OPTIONS
Response
OPTIONS
Response
OPTIONS
Request times-out
Requests sent to Backup
OPTIONS
Request times-out
OPTIONS
Response
Requests sent to Primary

13. SIP Authentication

Overview

The gateway supports the use of SIP Authentication. SIP Authentication is based upon HTTP Authentication as specified in RFC 2617. SIP Authentication follows RFC 2617 closely with a few exceptions which are noted in section 22 of RFC 3261. One notable exception is the lack of Basic Authentication in SIP as opposed to HTTP. SIP MUST use Digest authentication. RFC 2617 specifies two types of authentication: Basic and Digest. Basic authentication sends the user password response as clear text while digest authentication encodes the (checksum) response using MD5 hashing. Basic authentication is not supported in SIPv2 (RFC 3261) because it is deemed too insecure. Therefore only Digest Authentication is implemented int the gateway.

In the challenge scheme, a UAS may respond to an incoming method with a 401 or 407 response which contains a "realm". The UAC must know about the realm and have a valid username and password associated with the realm of the UAS. The UAC then re-sends the method with the realm, username and an encrypted version of the password. Upon receiving the updated method, the UAS authenticates that the username is a valid and the password matches. If this test passes then the UAS proceeds with SIP communication as normal. If the authentication procedure fails, the UAS may respond with a 406 and terminate SIP communication.

The gateway allows for Authentication to be enabled or disabled on a per method basis. Also, authentication may be enabled or disabled for the gateway acting as a SIP client (UAC) or a SIP server (UAS). The INVITE, REGISTER , NOTIFY, INFO, BYE, REFER and OPTIONS methods may all be controlled individually with regard to SIP Authentication. Per RFC 3261, the CANCEL and ACK methods must NOT be challenged. The CANCEL and ACK methods will however carry the same authentication information which appeared in their associated INVITE method.

The term "Authenticate" is used when a UAS challenges an incoming method with a 401 response and realm. The term "Authorization" is used when the UAC re-sends the method with the username associated with the realm and the encrypted password credentials.

The most common uses of SIP Authentication are detailed in diagrams later in this section.

Example WWW-Authenticate Header

The following shows an example '401 Unauthorized' response to an original INVITE. When the UAS requires authentication, it sends this response as opposed to a 200OK and includes the WWW-Authenticate header (if the UAS is a proxy it will send the 'Proxy-Authenticate' header). This header indicates how the UAC must respond in order to be authenticated to request service from the UAS.

This example shows a WWW-Authenticate header with a realm='PIMG21'. The UAC must have knowledge of this realm and must have a valid username and password for the realm. The header also specifies that Digest authentication must be used and that the encryption algorithm to be employed is MD5. For a complete description of the WWW-Authenticate header please refer to RFC 2617.

```
SIP/2.0 401 Unauthorized
WWW-Authenticate: Digest
realm="PIMG21",nonce="c081cbc58e2420e33dc34c2ea98431f6",opaque="86471f3c03
553c7098a320394a941973",algorithm=MD5,qop="auth"
From:<sip:101@192.168.1.20:5060;user=phone>;vnd.pimg.port=1;tag=2744324631353641000
D3B4C
To:<sip:102@192.168.1.21;user=phone>;tag=3BEC324631353641000D3973
Call-ID:01B2285F4481400000000013@englabocs.local.Gw62
CSeq:1 INVITE
Server:PBX-IP Media Gateway/2.1
Via:SIP/2.0/UDP 192.168.1.20:5060;branch=z9hG4bKC0EA65ED138A3CBFF9605A0F931EDF59
Content-Length:0
```

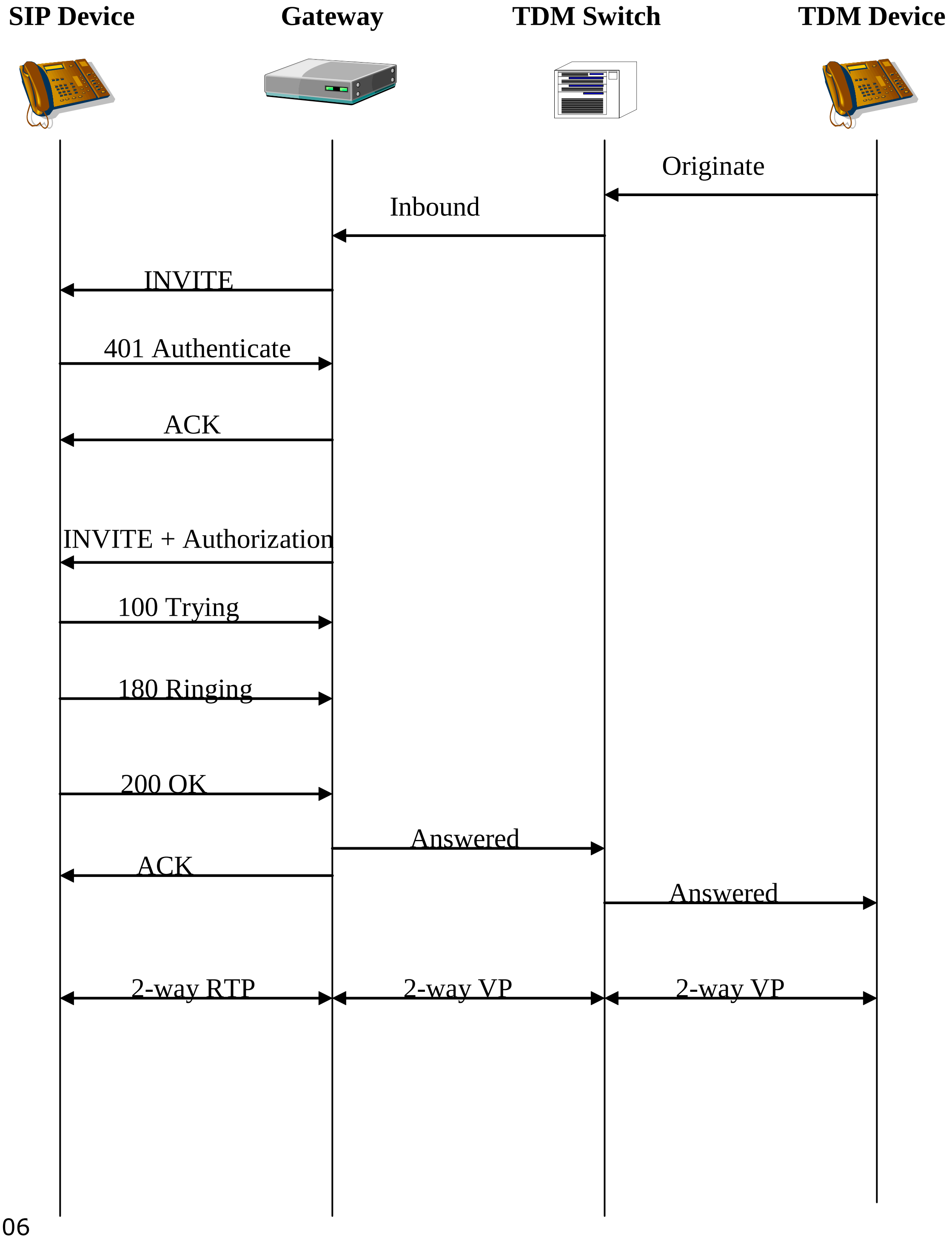
Example Authorization Header

The following INVITE shows a UAC attempting a re-send of the original INVITE after having received a 401 Unauthorized challenge. The UAC inserts the ‘Authorization’ header which was not present in the original request. The UAC performs a database lookup of known devices and finds the ‘PIMG21’ realm. The UAC has a username=’GatewayUser’ and a password associated with the PIMG21 realm. The encrypted value of the password is placed in the response and the INVITE is re-sent to the UAS.

```
INVITE sip:102@192.168.1.21;user=phone SIP/2.0
From:<sip:101@192.168.1.20:5060;user=phone>;vnd.pimg.port=1;tag=0611324631353641000
D3C44
To:<sip:102@192.168.1.21;user=phone>
Content-Type:application/sdp
Supported:replaces,100rel
Authorization:Digest
username="GatewayUser",realm="PIMG21",nonce="c494e9bf62649e20b0e772dd37f0
9000",response="8daf846e1233a845db05a4cc3fc48036",uri="sip:102@192.168.1.21;u
ser=phone",cnonce="a8d641f1d9238e1be78de6a9cb67d0d7",opaque="28c4fcb3354ea
f9dc3de5b73215ccf09",qop=auth,algorithm=MD5,nc=00000001
Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,PRACK,REFER,SUBSCRIBE,NOTI
FY,MESSAGE
Expires:120
Call-ID:01B2285F5D81400000000015@englabocs.local.Gw62
CSeq:1 INVITE
Max-Forwards:70
User-Agent:PBX-IP Media Gateway
Contact:sip:101@192.168.1.20:5060
Prot      Via:SIP/2.0/UDP
192.168.1.20:5060;branch=z9hG4bK5BE0968A5F6B47E7888B139EDE3BCE47
Content-Length:232
```

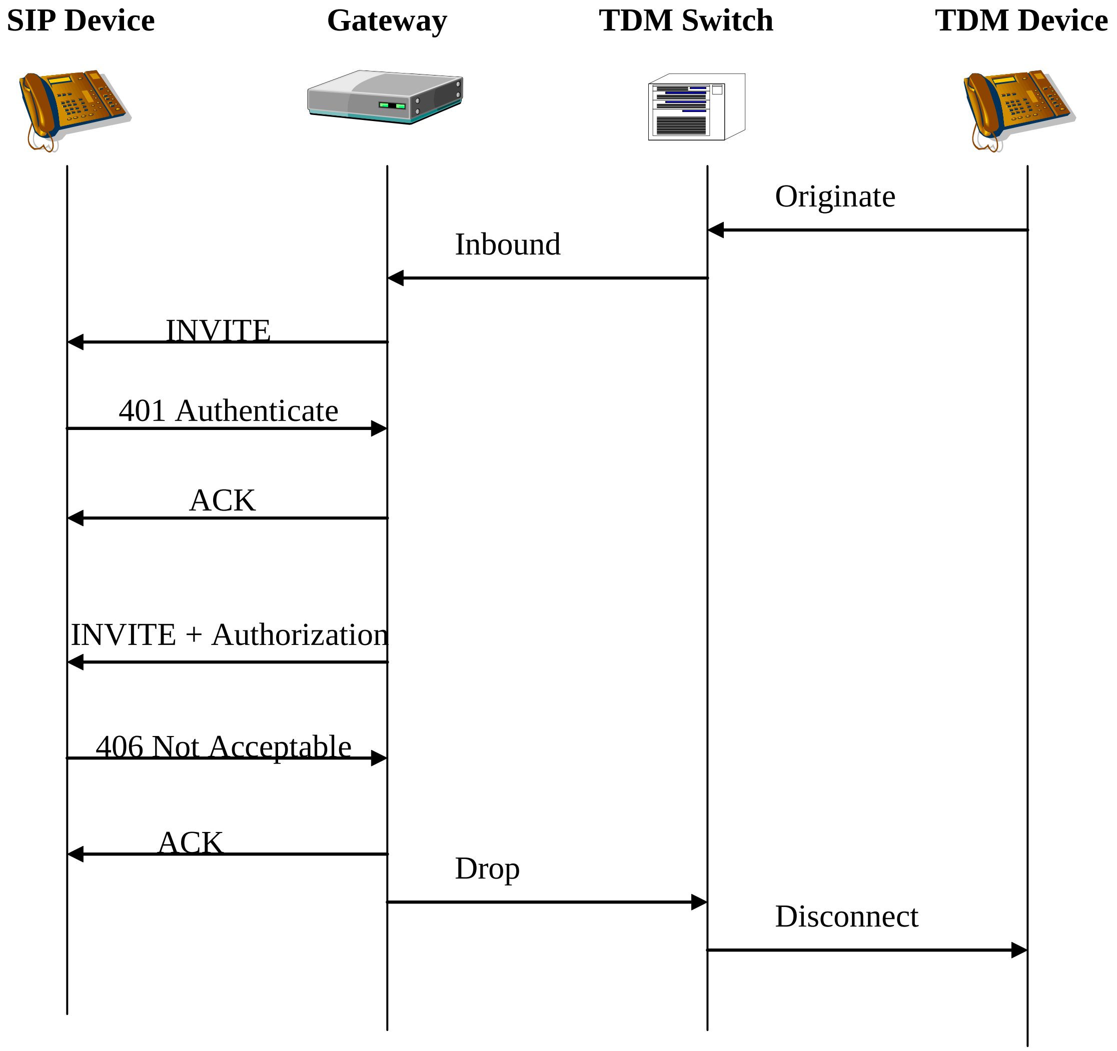
INVITE Authentication Success – Gateway is UAC

The following shows the gateway acting as the client and sending an INVITE method to a SIP server device which requires authentication for the INVITE method. The SIP device challenges the gateway with a 401 Authentication message in response to the original INVITE. The gateway re-sends the INVITE with an Authorization header based upon data that was enclosed in the 401 response. The SIP device Authenticates the new INVITE and the call proceeds as normal.



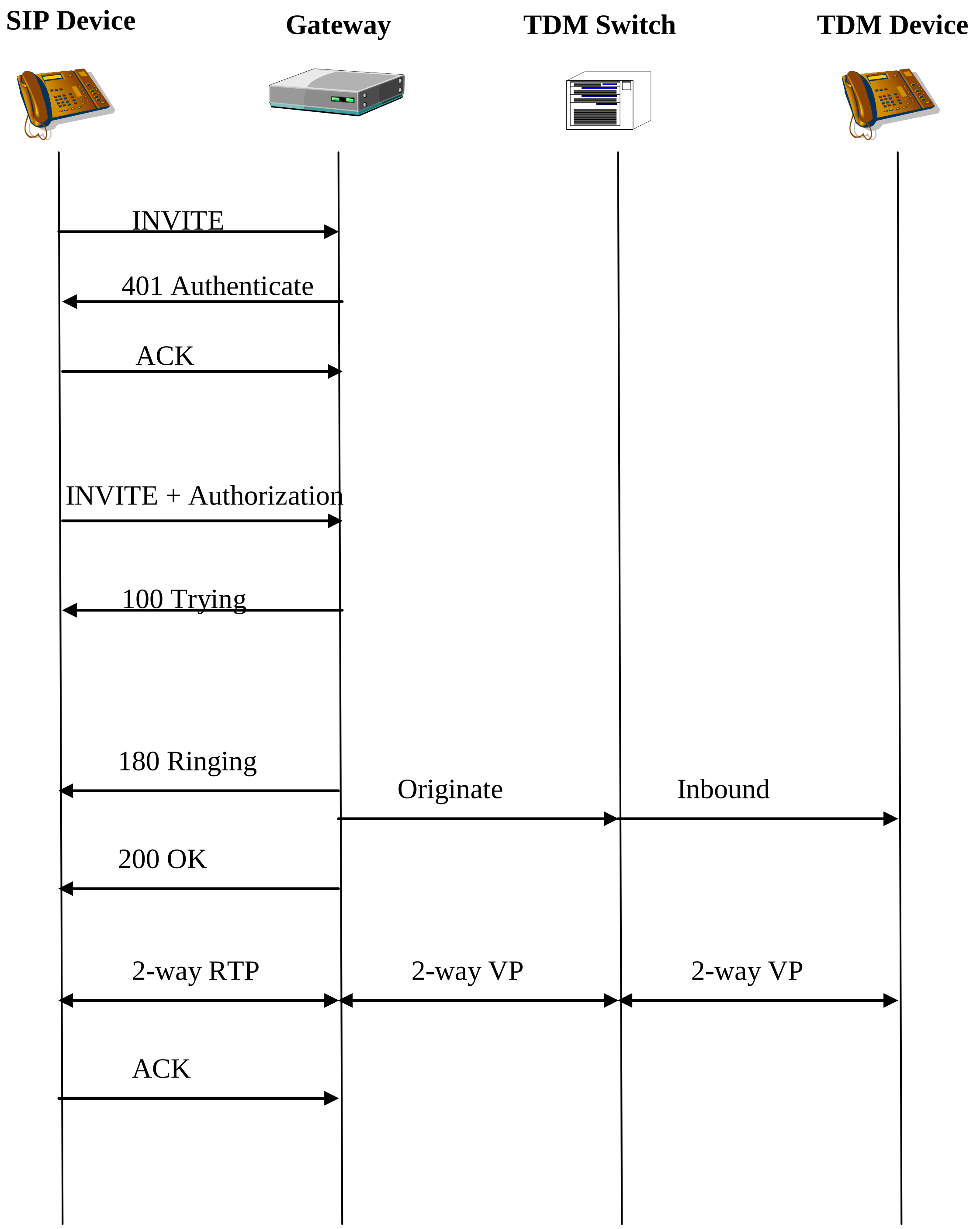
INVITE Authentication Failure – Gateway is UAC

This message exchange depicts a case where the gateway is challenged by SIP device. In this exchange, the gateway   re-sends the INVITE message with an Authorization header. The SIP device examines the Authorization header and subsequently rejects the INVITE. The rejection may be due to an invalid username or invalid encrypted password.



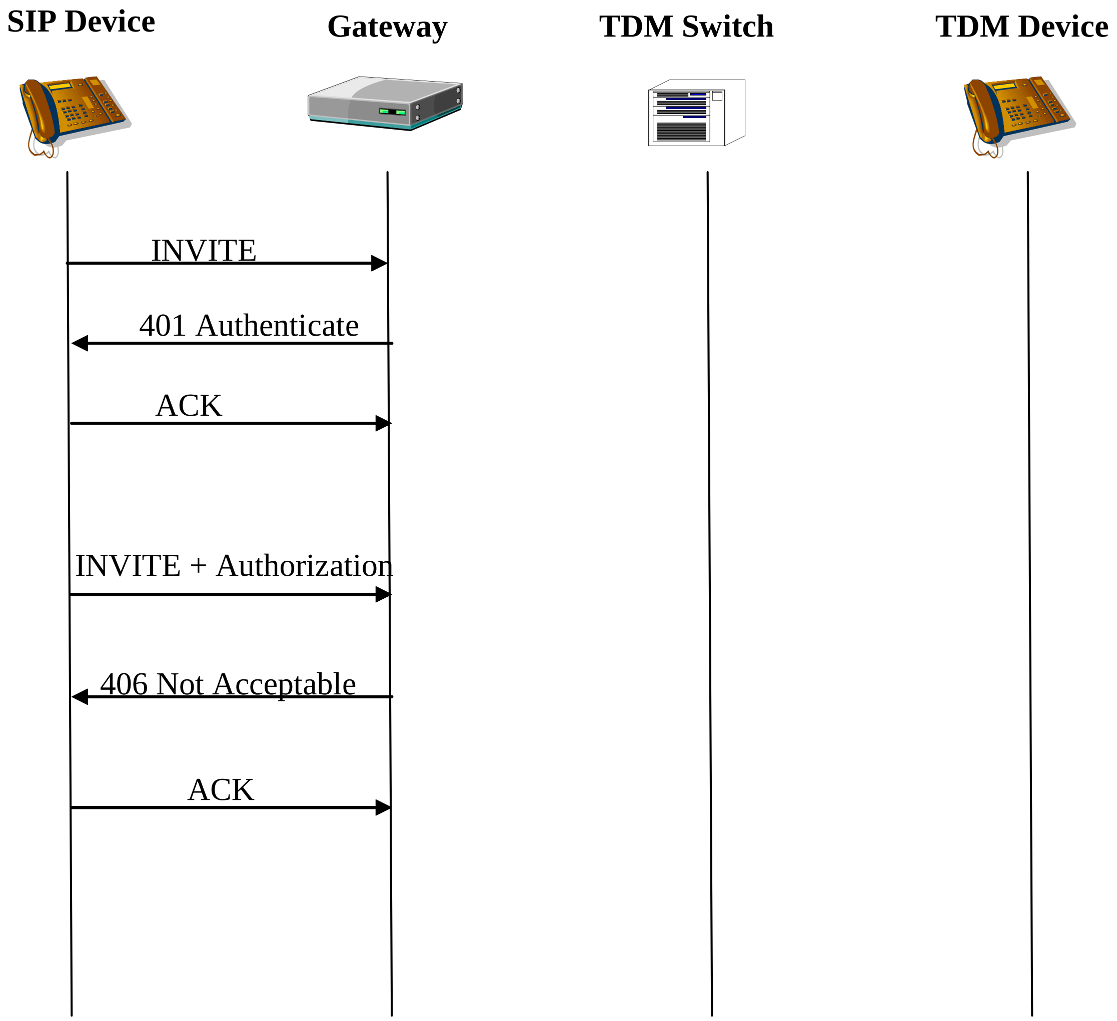
INVITE Authentication Success – Gateway is UAS

The following shows the gateway acting as the server and receiving an INVITE method. With Authentication enabled on the UAS side, the gateway challenges the client with a 401 Authenticate message in response to the original INVITE. The SIP Device client then re-sends the INVITE with the Authorization header based upon data that was enclosed in the 401 response. The gateway authenticates the new INVITE and the call proceeds as normal.



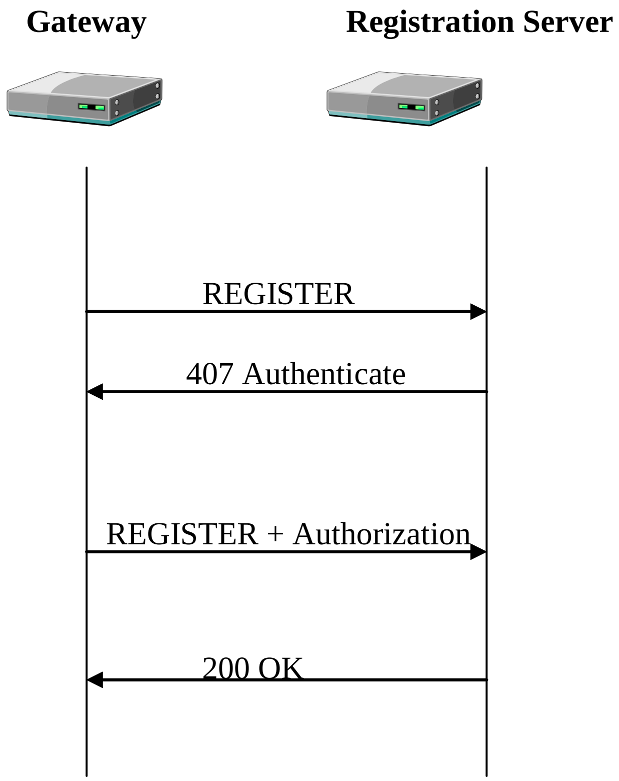
INVITE Authentication Failure – Gateway is UAS

This message exchange depicts a case where the gateway challenges a SIP device. In this exchange, the SIP Device re-sends the INVITE message with an Authorization header. The gateway examines the Authorization header and subsequently rejects the INVITE. The rejection may be due to an invalid username or invalid encrypted password.



Register Authentication Success – Gateway is UAC

A typical Authentication use case is the gateway acting as a client and attempting to register itself with a registration server. Registration servers often require SIP Authentication. If the Registration server also acts as a Proxy server it may challenge the REGISTER method with a 407 Authenticate as opposed to a 401 response. The gateway is capable of responding properly to a 401 or a 407 response from a UAS. Here the gateway attempts to register its presence and is challenged with a 407 response. The gateway reads the realm from the 407 response and re-sends the REGISTER with the username and password credentials associated with the realm of the registration server.



14. Calling ID Privacy

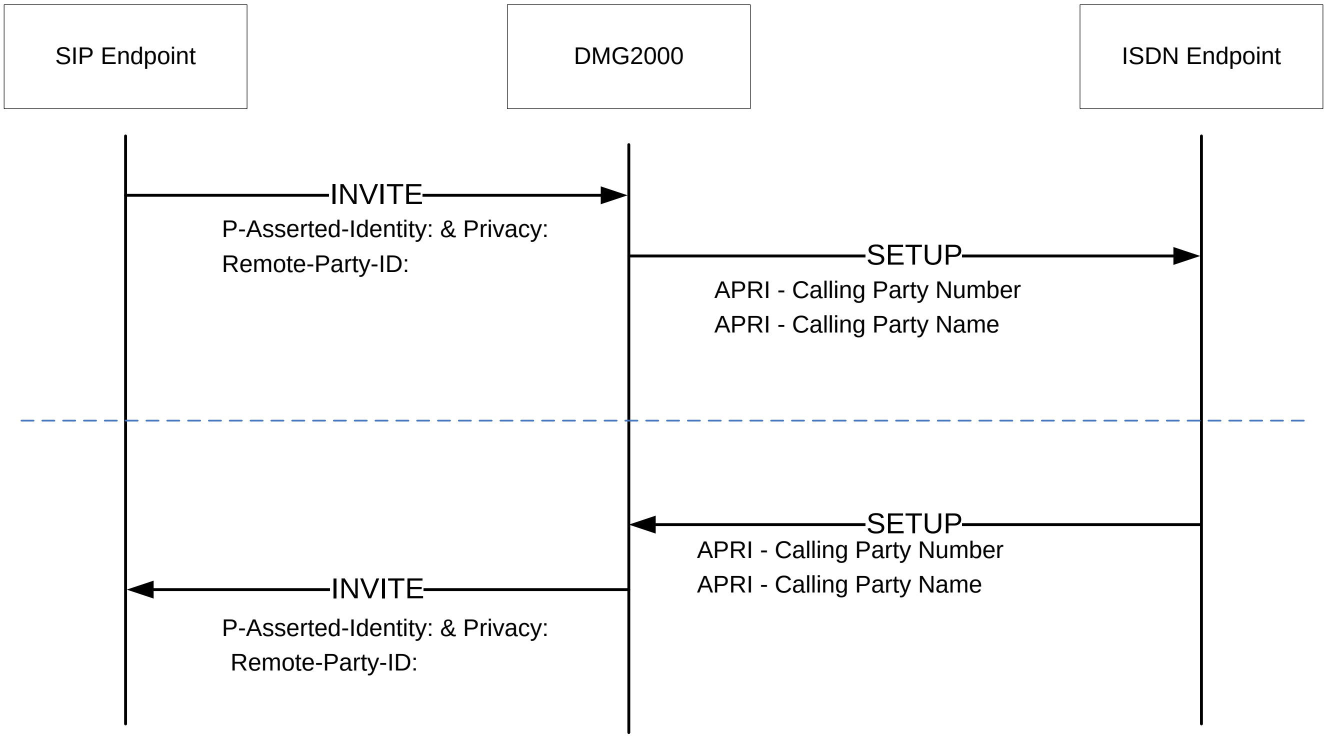
The DMG2000 series gateways support the privacy of Calling ID between SIP and ISDN endpoints. This is accomplished by translating between SIP privacy headers and ISDN APRI (Address Presentation Restriction Indicator) bits.

On the SIP side there are three headers associated with Calling ID privacy. They are:

- P-Asserted-Identity & Privacy (always appear together)
- Remote-Party-ID

On the ISDN side there is an APRI bit associated with the Calling Party Number and one with the Calling Party Name.

A high level graphical view of the translation is shown following. The top half of the drawing shows a SIP INVITE message containing the three privacy headers being translated into an ISDN SETUP message that contains the APRI bits for the calling party number and name. The drawing's lower half shows the reverse direction where a ISDN SETUP message is translated into a SIP INVITE message.


SIP Endpoint
DMG2000
ISDN Endpoint
INVITE
P-Asserted-Identity: & Privacy:
Remote-Party-ID:
SETUP
APRI - Calling Party Number
APRI - Calling Party Name
SETUP
APRI - Calling Party Number
APRI - Calling Party Name
INVITE
P-Asserted-Identity: & Privacy:
Remote-Party-ID:

Gateway Configuration

The gateway can be configured to enable or disable translation. When translation is enabled the gateway is configured to translate only ‘P-Asserted-Identity & Privacy’, only ‘Remote-Party-ID’ or to translate both.

Translation

The following subsections present the gateway's translation between SIP Privacy headers and ISDN APRI bits.

SIP INVITE to ISDN SETUP

Privacy Header Method: P-Asserted-Identity		
Input	Output	
Privacy:	APRI - Calling Party Number	APRI – Calling Party Name
id	restricted	restricted
header	restricted	restricted
user	restricted	restricted
none	allowed	allowed

Privacy Header Method: Remote-Part-ID		
Input	Output	
Remote-Party-ID:	APRI - Calling Party Number	APRI – Calling Party Name
full	restricted	restricted
name	allowed	restricted
uri	restricted	allowed
off	allowed	allowed

Privacy Header Method: Both			
Input		Output	
Remote-Party-ID:	Privacy:	APRI - Calling Party Number	APRI – Calling Party Name
full	don’t care	restricted	restricted
name	don’t care	allowed	restricted
uri	don’t care	restricted	allowed
off	don’t care	allowed	allowed
<none>[1]	id	restricted	restricted
<none>	header	restricted	restricted
<none>	user	restricted	restricted
<none>	none	allowed	allowed

[1] <none> means that the Remote-Party-ID header is not in the SIP INVITE message.

ISDN SETUP to SIP INVITE

Privacy Header Method: P-Asserted-Identity		
Input		Output
APRI – Calling Party Number	APRI – Calling Party Name	Privacy:
allowed	allowed	none
allowed	restricted	none
restricted	allowed	id
restricted	restricted	id

Privacy Header Method: Remote-Party-ID		
Input		Output
APRI – Calling Party Number	APRI – Calling Party Name	Remote-Party-ID:
allowed	allowed	off
allowed	restricted	name
restricted	allowed	uri
restricted	restricted	full

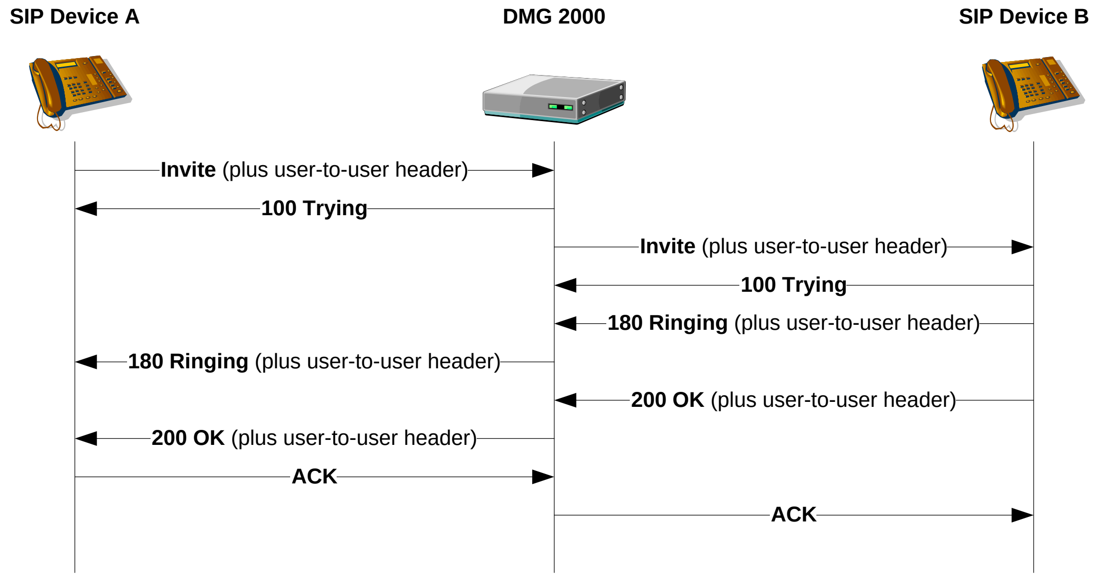
Privacy Header Method: Both			
Input		Output	
APRI – Calling Party Number	APRI – Calling Party Name	Privacy:	Remote-Party-ID
allowed	allowed	none	off
allowed	restricted	none	name
restricted	allowed	id	uri
restricted	restricted	id	full

15. User To User Information

The **DMG2000** series gateways support the transporting of UUI (User to User information) using a SIP header as described in "draft-johnston-cuss-sip-uu1-01.pdf" section 3.6. The gateway supports UUI information in the INVITE method, and the 180 Ringing and 200 OK responses. Also supported SIP REFER when used for transfers.

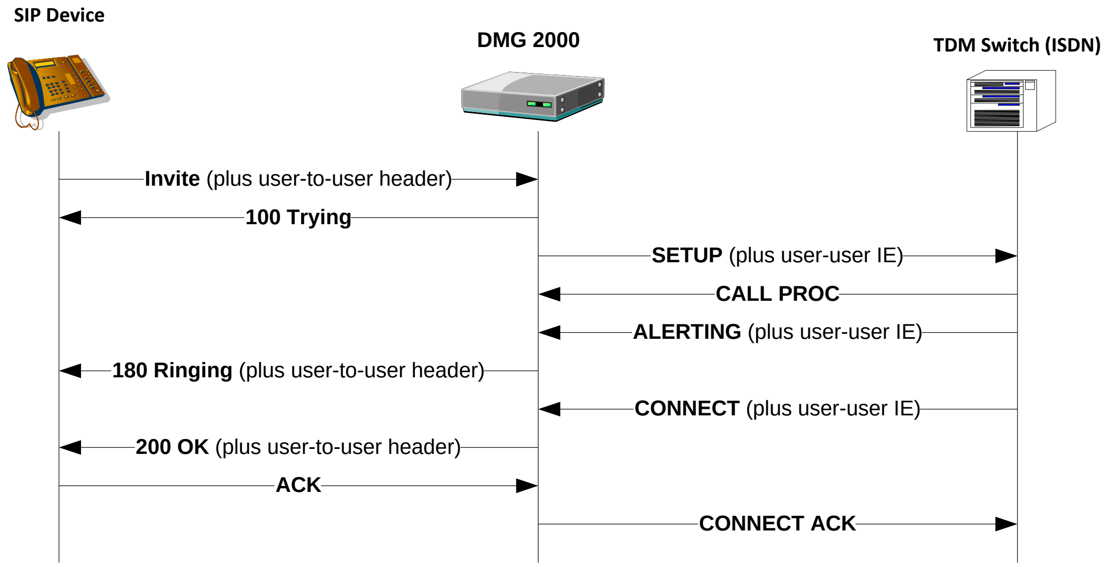
SIP to SIP calls

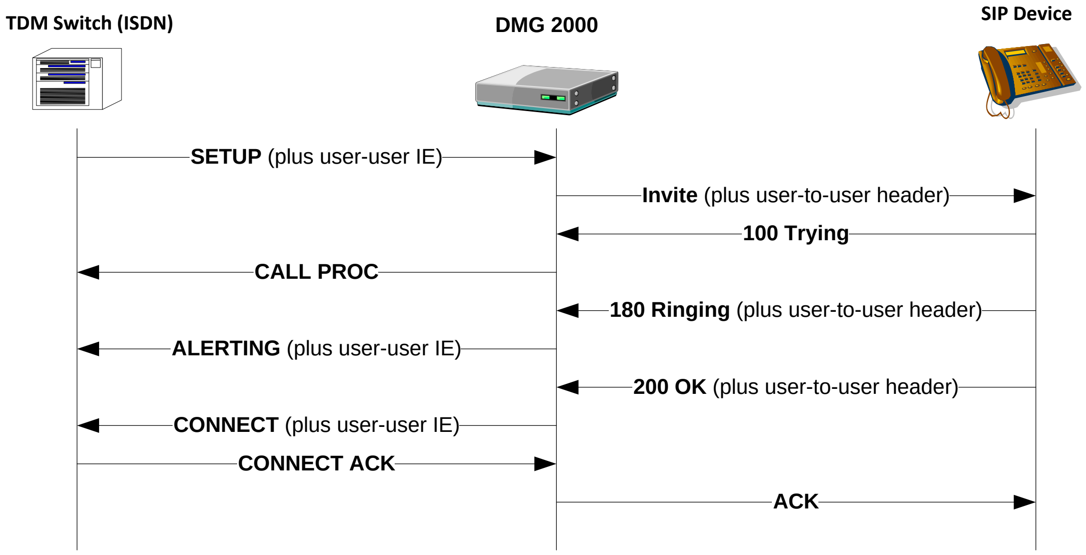
For SIP to SIP calls the user-to-user header is passed unchanged from inbound call to outbound call.


SIP Device A
DMG 2000
SIP Device B
Invite (plus user-to-user header)
100 Trying
Invite (plus user-to-user header)
100 Trying
180 Ringing (plus user-to-user header)
180 Ringing (plus user-to-user header)
200 OK (plus user-to-user header)
200 OK (plus user-to-user header)
ACK
ACK

SIP to TDM and TDM to SIP calls

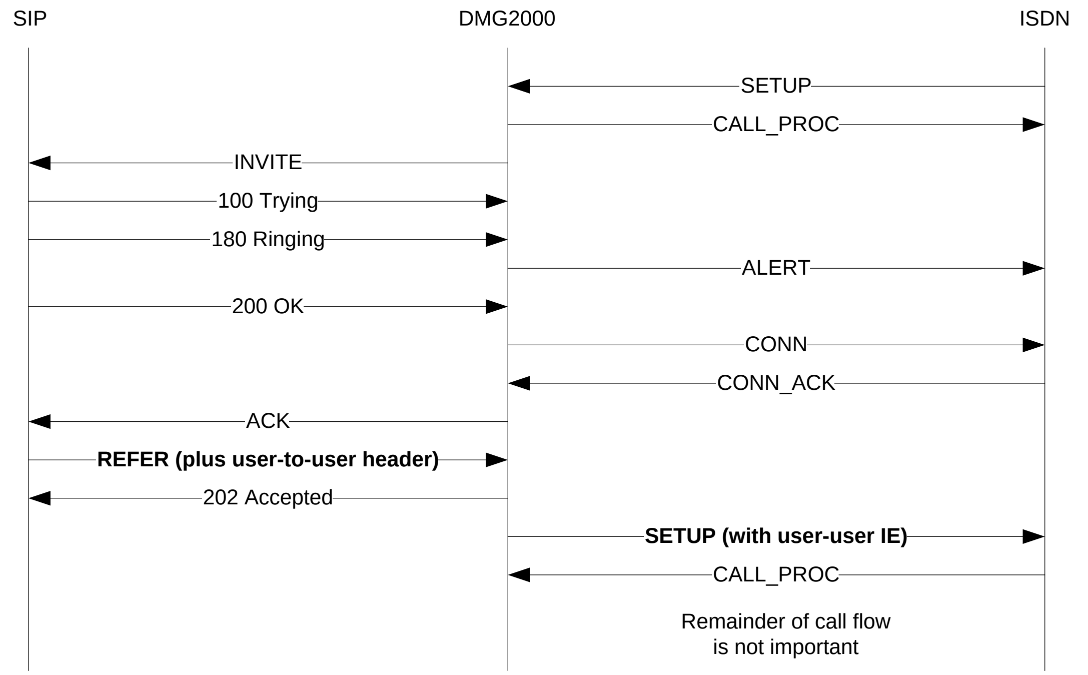
For calls between SIP and TDM an inbound user-to-user header is translated into an outbound ISDN user-user information element, and an inbound ISDN user-user information element is translated into an outbound user-to-user header.

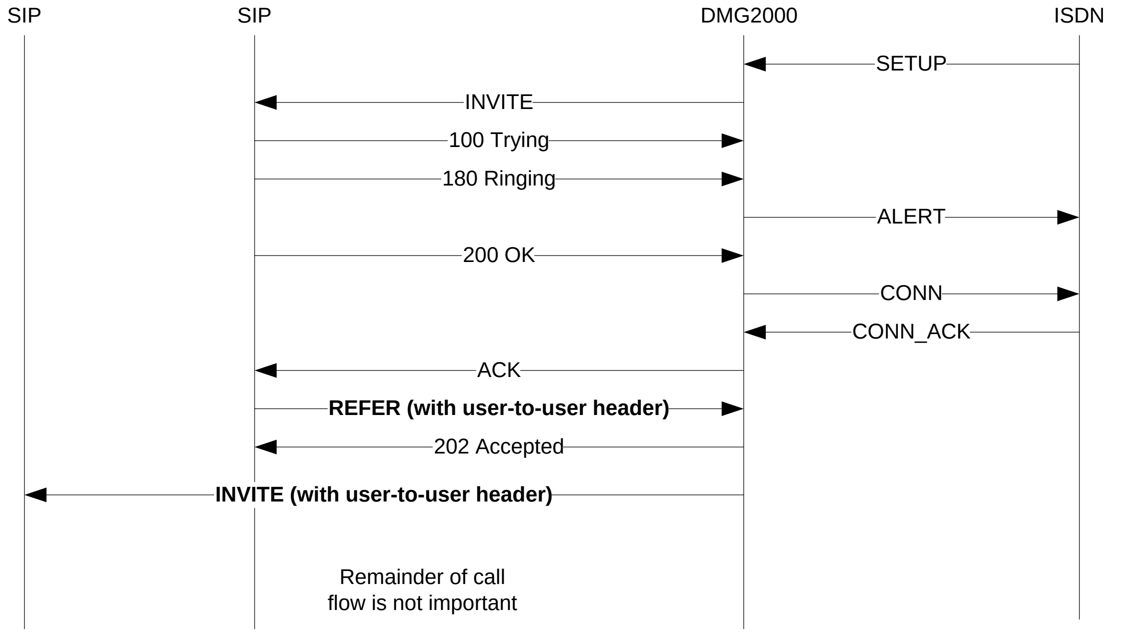

SIP Device
DMG 2000
TDM Switch (ISDN)
Invite (plus user-to-user header)
100 Trying
SETUP (plus user-user IE)
CALL PROC
ALERTING (plus user-user IE)
180 Ringing (plus user-to-user header)
CONNECT (plus user-user IE)
200 OK (plus user-to-user header)
ACK
CONNECT ACK


TDM Switch (ISDN)
DMG 2000
SIP Device
SETUP (plus user-user IE)
Invite (plus user-to-user header)
100 Trying
CALL PROC
180 Ringing (plus user-to-user header)
ALERTING (plus user-user IE)
200 OK (plus user-to-user header)
CONNECT (plus user-user IE)
CONNECT ACK
ACK

Transfers using SIP REFER

When the DMG2000 receives a SIP REFER that contains UUI, the UUI is passed to the resulting SIP INVITE or ISDN SETUP. Supported call flows are as follows:





Supported Encoding

The gateway supports two types of encoding in a user-user header: ‘string’ or ‘hex’.

String Encoding

String encoding uses a single ascii character to encode a single byte of data. For example, an ascii ‘U’ translates into the byte 55h. An ascii ‘b’ translates into the byte 62h.

Hex Encoding

Hex encoding uses two ascii digits to encode a single byte of data. For example, the two ascii digits ‘87’ translate into a single byte of 87h.

Comparison of String verse Hex Encoding

The following table lists some example user-to-user headers and the corresponding ISDN data

User-to-User Header	ISDN Data
User-to-User:63616CC0; encoding=hex	translates to four bytes: 63h 61h 6Ch C0h
User-to-User:63616CC0; encoding=string	translates to eight bytes: 36h 33h 36h 31h 36h 43h 43h 30h
User-to-User:Five Days; encoding=string	translates to nine bytes: 46h 69h 76h 65h 20h 44h 61h 79h 73h
User-toUser:Five Days; encoding=hex	no translation as the data is not ascii digits

16. Multipart MIME to encapsulate PSTN signaling

The DMG2000 series gateways support encapsulation of PSTN signaling using multipart MIME bodies as presented in RFC 3204. This section describes the various PSTN signaling elements that the DMG2000 supports.

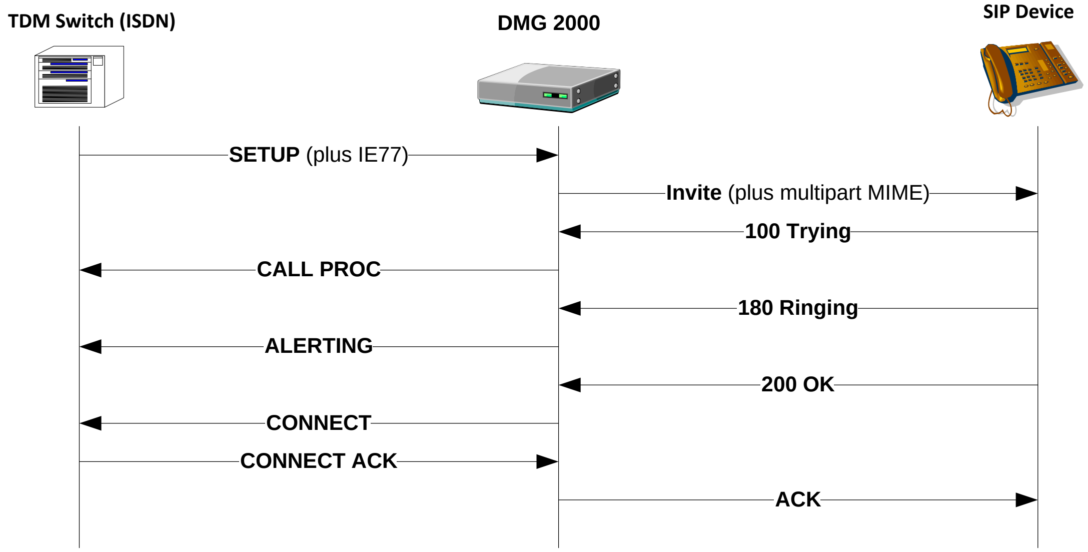
ISDN Information Elements

Information Element 77 (hex)

Information Element 77 (hex) is used by France Telecom to send proprietary information. The format of the information element is as follows:

8	7	6	5	4	3	2	1	
0	1	1	1	0	1	1	1	byte 1
Length of contents of information elements (in bytes)								byte 2
Contents of information element								byte 3...N

When IE77 is received in an ISDN SETUP message, the gateway propagates the information element to a SIP INVITE message. If the gwIsdnIe77Enable configuration variable is set to YES.



The gateway uses a multipart MIME body to encapsulate the data. The following is an example of this encapsulation.

```
1       INVITE sip:4700@10.10.11.47 SIP/2.0
2       From:<sip:2042@10.10.11.114:5060;user=phone>;vnd.pimg.port=23;tag=5A533246
3       To:<sip:4700@10.10.11.47>
4       Contact:<sip:2042@10.10.11.114:5060>
5       Mime-Version: 1.0
6       Content-Type:multipart/mixed ;boundary="?(boundary)*"
7       Supported:replaces,early-session,100rel
8       Allow:INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,PRACK,REFER,NOTIFY
9       Expires:120
10      Call-ID:01B246A660AA00460000000E@pbxgw.default.com
11      CSeq:1 INVITE
12      Max-Forwards:70
13      User-Agent:PBX-IP Media Gateway
14      Via:SIP/2.0/UDP 10.10.11.114:5060;branch=z9hG4bKE194EDF
15      Content-Length:291
16
17      --?(boundary)*
18      Content-Type: application/SDP
19       v=0
20      o=phone 15122 16488 IN IP4 10.10.11.114
21      s=-
22      c=IN IP4 10.10.11.114
23      t=0 0
24      m=audio 49030 RTP/AVP 0 13
25      a=rtpmap:0 PCMU/8000/1
26      a=ptime:30
27      a=rtpmap:13 CN/8000
28
29      --?(boundary)*
30      Content-Type: application/ISUP; version=Dialogic-DMG2000-v1
31      Content-Disposition: signal ; handling=optional
32      Content-Description: ISDN IE 77h
33
34      37 31 36 35 35 35 36 35 35 32
35
36      --?(boundary)*--
```

Lines 5 and 6 specify that multipart MIME is appended to the INVITE message and that it uses MIME version 1.0. Line 6 defines the boundary between MIME bodies; in our example the boundary is the string **?(boundary)***.

Line 17 starts the first MIME body. This body sends SDP (session description protocol) to the SIP endpoint. We are not concerned with this; it is described in section 0.

Line 29 starts the second MIME body. This body sends the ISDN information element's data. Lines 30-32 describe the body to allow the SIP endpoint to decode it. Line 30 states the MIME body version. Line 31 says that processing the MIME body is optional. This allows for the case when some SIP endpoints need the ISDN information element but others don't. Line 32 specifies the content of the body is an ISDN information element number 77 (hex). Each data byte of the information element is translated into two hex digits on line 34. Line 36 simultaneously ends the second body and the multipart MIME.