

IP-based Voice Mail

MediaPack™ & Mediant™ Series Analog and Digital VoIP Media Gateways

Version 7.2

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

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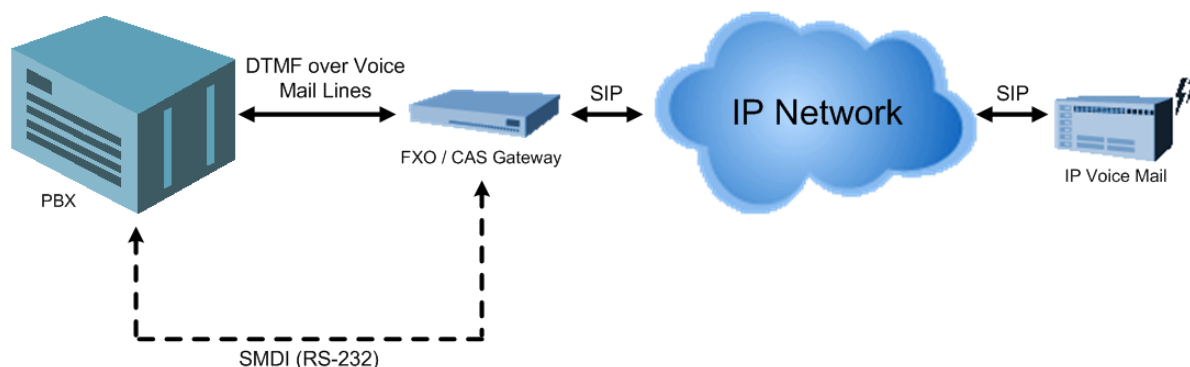
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1 Overview

The gateway can be used to mediate between a PBX and an IP Voice Mail (VM) server. The supported architecture (illustrated in [Figure 1-1](#)) includes an AudioCodes gateway connected to a PBX using voice mail lines (FXO, CAS, or QSIG interface), and connected to an IP VM via the IP network. The PBX is unaware of the gateway that is utilized between it and the IP VM, and operates as if it is connected directly to the VM.

The gateway communicates with the PBX by using either QSIG or Simplified Message Desk Interface (SMDI) via the serial RS-232 connection (refer to [Section 2](#) on page 9), or special in-band DTMF digit patterns (refer to [Section 3](#) on page 29).

Figure 1-1: Supported Voice Mail Architecture for FXO/CAS Gateways



1.1 Supported Features

The following interworking features are supported:

- **Forward Calls:** delivery of VM messages to the IP VM (PBX→gateway (GW)→VM).
- **Direct Call:** retrieval of VM messages from the IP VM (PBX→GW→VM).
- **Message Waiting Indication:** notifying the PBX on remaining VM messages (VM→GW→PBX).
- **Call Transfer:** transferring calls to an operator or to a different PBX extension (VM→GW→PBX).
- **PBX Disconnect Supervision:** detected by several methods: current disconnect (FXO), Busy/Reorder tones, and DTMF Code or Silence detection. After Disconnect detection, the gateway terminates its session with the PBX and VM.
- **Outgoing Call:** call to PBX subscriber, initiated by VM.

1.2 Supported Products

This document is applicable to the following devices:

- Mediant 5xx series:
 - Providing FXO interfaces
 - Configured with Channel Associated Signaling (CAS) and QSIG protocols
- Mediant 8xx series:
 - Providing FXO interfaces
 - Configured with Channel Associated Signaling (CAS) and QSIG protocols
- Mediant 1000 series:
 - Providing FXO interfaces
 - Configured with Channel Associated Signaling (CAS) and QSIG protocols

Throughout this document, unless otherwise specified:

- Whenever reference is made to CAS/ISDN with Simplified Message Desk Interface (SMDI), it applies to the following products:
 - Mediant 5xx series
 - Mediant 8xx series
 - Mediant 1000 series
- Whenever reference is made to analog lines with SMDI, or to in-band DTMF digit patterns, it applies to the following products:
 - Mediant 5xx series
 - Mediant 8xx series
 - Mediant 1000 series

2 SMDI Operation

This section describes the SMDI operation.

2.1 BellCore SMDI

The BellCore SMDI provides the gateway with data, enabling it to intelligently process incoming voice-messaging calls. The SMDI protocol is defined in Telcordia GR-283-CORE standard. When routing a call (to the VM), the PBX sends an SMDI message to the gateway (through an RS-232 connection) informing it of the line it is using, the type of call it is forwarding, and information about the source and destination of the call.

Figure 2-1: SMDI Call Status Message Format

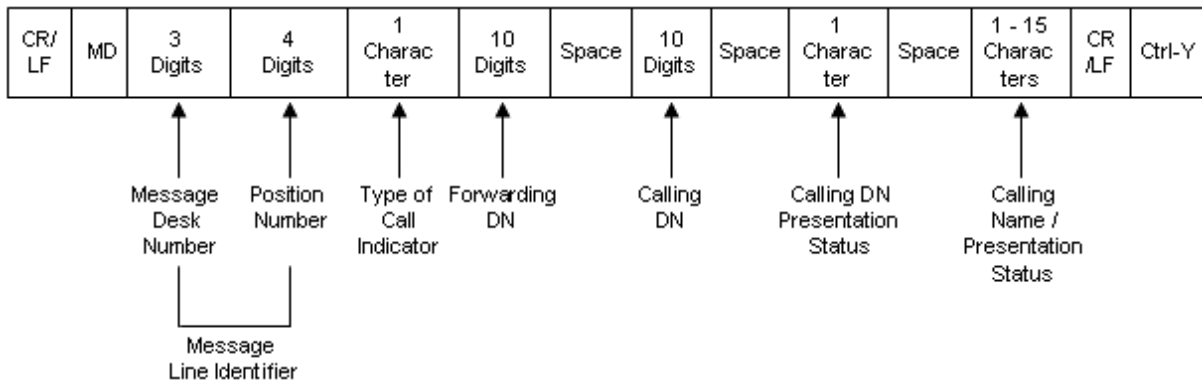
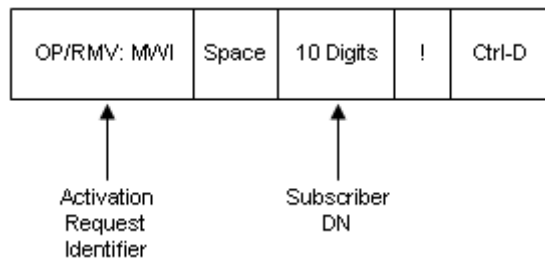


Figure 2-2: SMDI MWI Message Format



For BellCore SMDI examples, refer to Appendix A.

2.2 Ericsson MD-110 SMDI Variant

The Ericsson MD-110 PBX reports the call details to the voice mail system using RS-232 (V.24) interface. The protocol is proprietary and differs from BellCore SMDI standard. The V.24 signal is composed of one start bit, seven information bits, one parity bit (optional), and one stop bit. A data rate of 110 to 9600 bit/s is supported. It is possible to configure the PBX to use or not to use data flow checks (XON/XOFF-protocol).

The MD-110 PBX sends two messages for each forwarded call: one when the voice mail system answers the call and one when the calling party disconnects.

The signals exchanged between the voice mail system and the PBX consists of the following parts:

- **STX:** Start of text character.
- **V:** 2- to 5-digit voice mail port number (voice channel) that defines a voice channel input in a voice mail system. The 'V' field is similar to the Bellcore SMDI 'Message Line Identifier 7 digits' field.
- **T or D:** 2- to 10-digit directory number in the system. The 'T' field is similar to the Bellcore SMDI 'Forwarding DN' field. The 'D' field is similar to the Bellcore SMDI 'Calling DN' field.
- **SS:** 2-digit information system number that defines in which information system a message is waiting for the extension. (A '00' is reserved for the interception computer system.)
- **CR:** Carriage-return character.
- **LF:** Line-feed character.
- **PUB:** A-party number received from the public network (20 digits). The 'PUB' field is similar to Bellcore SMDI 'Calling DN' field for external call.

For example:

- **STX 80 T V CR LF:** Diverted call
Voice channel V has been seized for an incoming diverted call to an extension with directory number T.
- **STX 81 D V CR LF:** Direct internal call
Voice channel V has been seized for an incoming, direct internal call from an extension or PABX operator with directory number D. In the standard function, the voice mail system detects the call origin (internal or external) by the type of ring cadence.
- **STX 89 V CR LF:** Order to clear down the voice channel
The party that has had speech connection with the voice mail system on voice channel V has cleared down. The gateway releases the call if such a message is received.

The voice mail system can use the following messages to set Message Waiting Indications:

- **STX 06 T SS CR LF:** indicates that extension T has a message waiting in message system SS.
- **STX 07 T SS CR LF:** erase the message waiting indication for extension T and message system SS.

The MD-110 PBX sends heartbeat keepalive "STX 98 CR LF" messages. The voice mail system responds to this message with "STX 99 CR LF".

2.3 NEC SMDI Variant for NEAX 2400 IPX

The NEC PBX reports the call details to the voice mail system, using RS-232 (V.24) interface. The protocol is proprietary and differs from BellCore SMDI standard. The V.24 signal is composed of one start bit, seven information bits, one parity bit (optional), and two stop bits. A data rate of 1200 to 9600 bit/s is supported.

There are two message formats: ICS Format and IPX Format. Currently, only ICS format is supported. The extended Calling Party Information is also supported. If present in the message, it is used in the SIP From header instead of the "short" Calling Party number.

The general message format is shown in the figure below:

Figure 2-3: General Message Format

S T X	S A	U A	E I	Message Text (See CHAPTER 4.)	E T X
-------------	--------	--------	--------	----------------------------------	-------------

Legend:

- **STX:** Start of Text (ascii 2)
- **SA:** System Address (ascii 0)
- **UA:** Unit Address (ascii !)
- **EI:** Entry INDEX (ascii J)
- **ETX:** End of Text (ascii 3)

All messages begin with STX (ascii 2) and end with ETX (ascii 3).

CallHistory messages short format: 0!J001VVVVVV__CCI01SSSSSS001RRRRRR

- **V:** port number of the gateway to which the SMDI is referred. It is six chars padded with spaces, followed by two spaces. Usually, the port number contains four characters followed by two spaces.
- **CC:** call reason code that details the reason for the call being targeted to the Voice Mail (gateway):
 - **40:** Call forward on no answer
 - **41:** Call forward on busy
 - **42:** Call forward always (DND)
 - **43:** Direct call to voice mail
- **I:** 0 for internal call; 2 for external call.
- **SSSSSS:** source number (Calling number). It is six chars padded with spaces. For local station numbers, it usually contains four characters followed by two spaces. For external calls, it may contain three characters for route number and three for trunk number. Usually, for external call, this information is not relevant.
- **RRRRRR:** redirect number. It is six chars padded with spaces. For direct calls, it is all spaces because no Redirect number exists. For local station numbers, it usually contains four characters followed by two spaces.

The extension format can be identified by the character "A" at the end of the "short format" following the extension, as shown in Figure 2-4.

The Calling Number may be retrieved in three different ways (based on the received SMDI message arrived format):

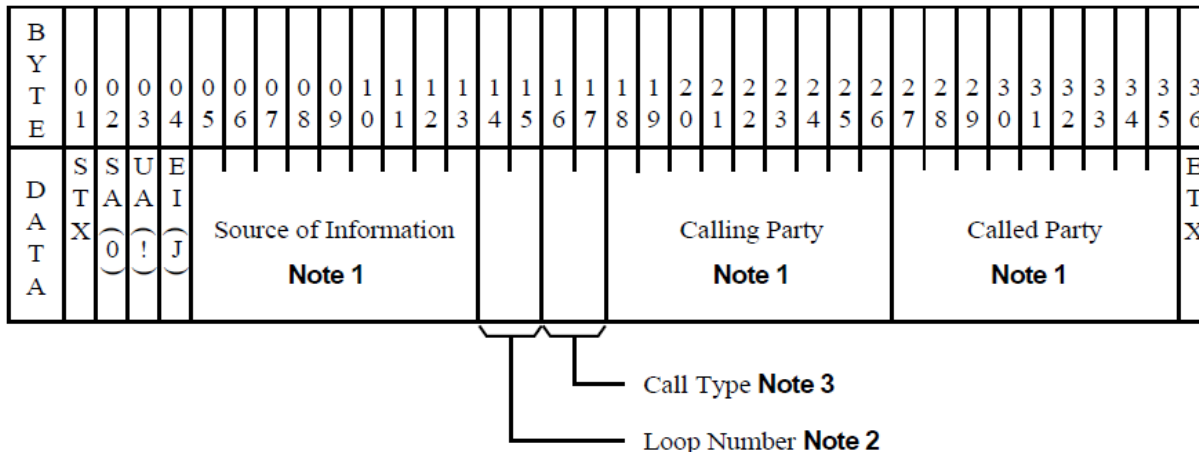
- Short format (no extension) - SSSSSS (bits 21-26) are used
- Extended format with Identifier 0 (Calling Number not provided) - Calling Number is obtained from bits 37-45 (based on Note 1)

- Extended format with Identifier 1 (Calling Number provided) - Calling Number is obtained from bits 47-78 (based on Note 1)

ICS Format: PBX to MC Message

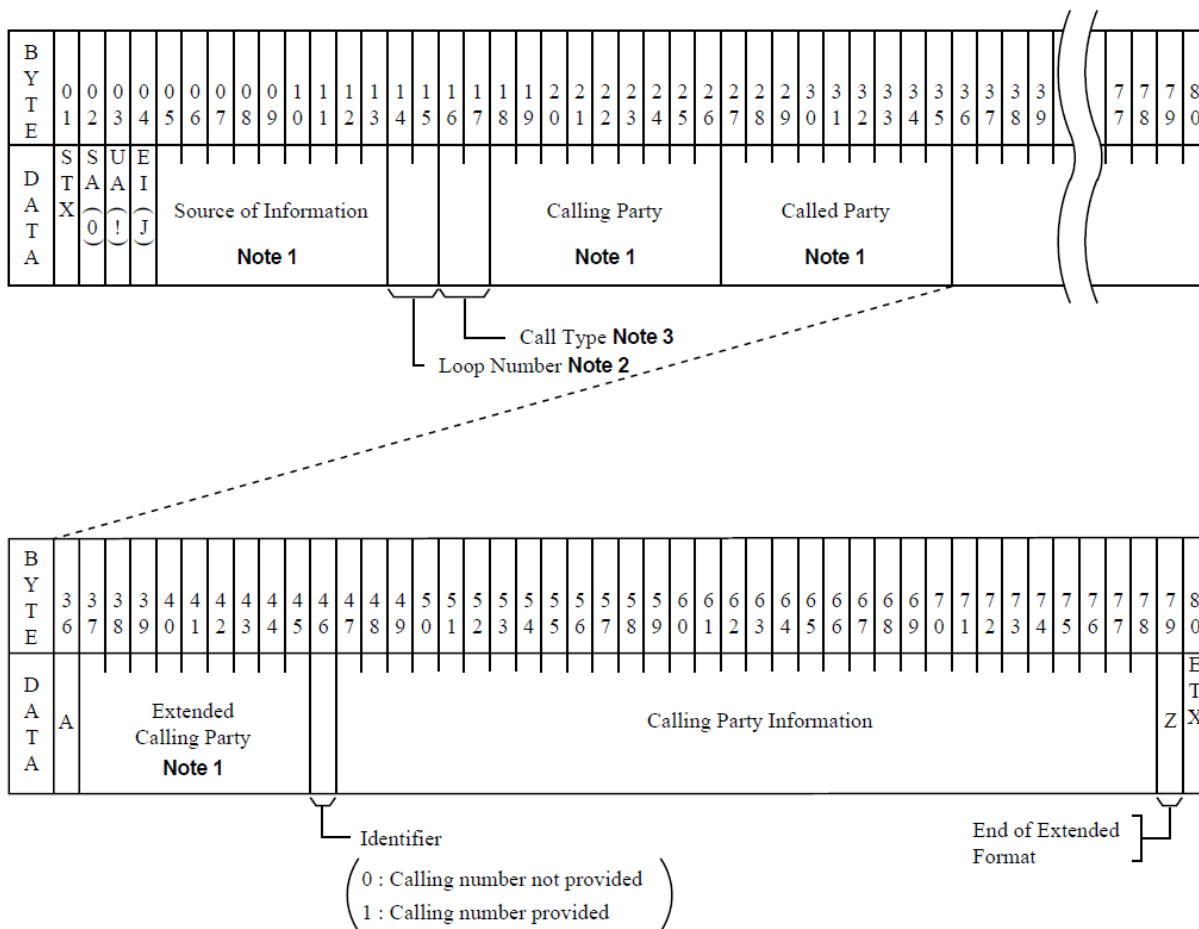
- When ASYD, INDEX 400, Bit 2 is "0", calling number information is not sent:

Figure 2-4: ICS Format - Bit 2 is "0"



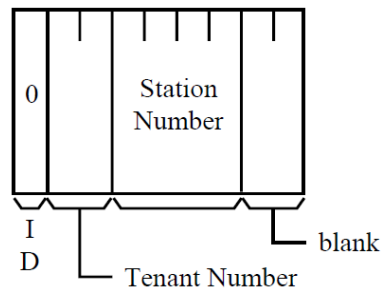
- When ASYD, INDEX 400, Bit 2 is "1", calling number information is sent:

Figure 2-5: ICS Format - Bit 2 is "1"

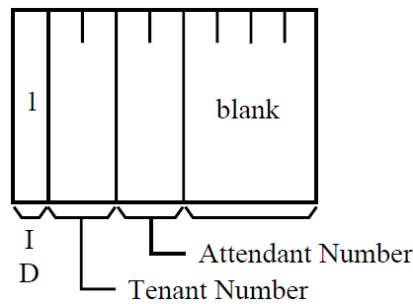


■ **Note 1:**

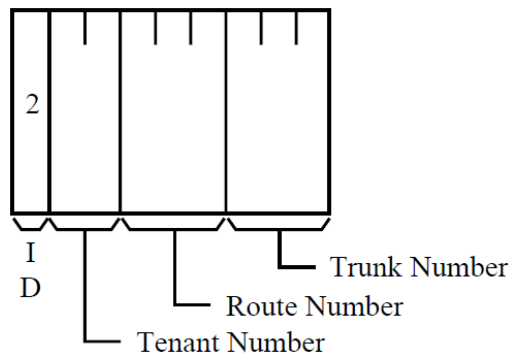
- When the destination is a station:



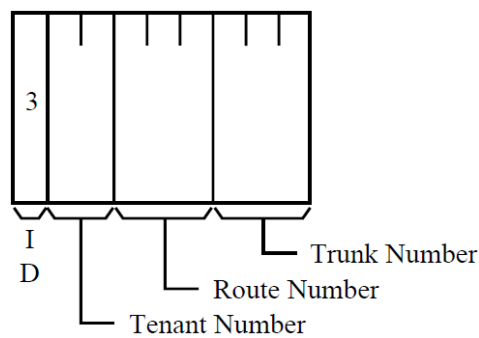
- When the destination is the Attendant:



- When the destination is a trunk:



- When the destination is a trunk (a calling number provided):



■ **Note 2:** This information is valid when Source of Information is the Attendant.

MsgWaiting messages format: 0!AMAAAAAABBBBBB

- **M:** 1 (or 2) = MwiOn; 5 (or 6) = MwiOff.
- **AAAAAA:** from Extension
- **BBBBBB:** to extension

The same “from” and “to” extension is used because SIP API can provide MWI for one endpoint, not for a range.

The SMDIMWINECFormat parameter defines the format of the MWI “M” values:

- 0 (default) for A1 and A5
- 1 for A2 and A6

The table below provides a few examples:

Table 2-1: Examples of NEC SMDI Variant for NEAX 2400 IPX

Example	Description
0!J0012216____420012201__0012200__	Call on port number 2216 was forwarded on DND. Call originated from caller 2201 to destination (redirect) 2200.
0!J0012001____43201005001001_____	Call received on port number 2001. Call originated from external PSTN (route:005, trunk:001), and is a direct call (the Redirect number is all spaces).
0!A12200__2200__	MWI ON for station 2200 (actually from 2200 to 2200).
0!A12200__2200__	MWI OFF for station 2200.

Below are examples of messages containing extended Calling Party Information:

- 0!J0017804 422011200090014175 A301120009**16613731616** Z
- 0!J00140530 4300122985 00120450 A 0 Z
- 0!J00140531 40201151002 00122105 A301 **15037267254** Z
- 0!J00140108 43201200011 00140100 A301200011 **12068526396** Z
- 0!J00140532 40201200006 00120190 A301200006 **12063548282** Z
- 0!J00140533 4300121245 00120450 A 0 Z

2.4 Calls from PBX to Voice Mail

The section describes calls from the PBX, through the gateway, to the voice mail.



Note: The description in this section applies to three SMDI types (i.e., BellCore, NEC NEAX 2400, and Ericsson MD-110), although the used terms are according to BellCore SMDI standard.

The gateway receives an incoming SMDI message from the PBX. This message is received at a short time interval before or after its corresponding call's setup. In this message, the PBX may include the Redirect Number (usually identifies the VM box number), Redirect Reason (forward reason), Source Number (if it doesn't exist, it is probably an external call to the VM), Source Number Presentation (Allowed / Restricted), and Source Name. Not all information is always available.

The SMDI Message Line Identifier (Message Desk Number and Position Number) field (seven characters number string) must match the gateway's local endpoint number. This endpoint handles the SMDI message and is expected to receive the call. Therefore, the local number of the gateway's endpoints must be correlated with SMDI messages identifiers.

The gateway seizes the PBX line immediately after detection of an incoming call. The FXO gateway performs this by using the Delayed Automatic Dialing feature. The CAS gateway performs this by using correct FXO Loop Start CAS table that is modified for SMDI applications. Seizing of the line is necessary to eliminate the possibility for glare condition of sending the voice mail message to the wrong voice mail inbox.

The gateway sends an INVITE message to the IP VM. If a Redirect Number is detected from the PBX's SMDI message, the INVITE message includes a Diversion header that contains the Redirect Number and Redirect Reason (according to IETF draft draft-levy-sip-diversion-08). This call is regarded as a 'forwarded call' (to the VM from the 'Redirect' extension) due to a Busy, No Answer, or Do Not Disturb (DND) condition (or some other unknown reason). It is also possible to configure the gateway to include "target" and "cause" parameters in the INVITE message according to RFC 4458 (Voicemail URI). This option is enabled by setting the *ini* file parameter EnableVMURI to 1.

The From header of the same SIP INVITE contains the extracted Source Number (if received in the SMDI message) as its User Part, and the Source Name. If a Source Number isn't received, the From header only contains the gateway's host name (unless a manipulation rule is used to add a calling number). The INVITE messages generated by the gateway contain Called Number as a User Part of the URI Request. For CAS and FXO gateways, this number can be preconfigured using the *ini* file.



Note: Sometimes the Called Number should be modified or removed using number manipulation rules.

To summarize: A Tel→IP call from a PBX to a VM via the gateway is processed as follows:

1. The Called Number is different for CAS and FXO gateways. The FXO gateway is typically configured to use delayed Automatic Dialing to a preconfigured number (using the TargetofChannel *ini* file parameter). This Called Number appears in the User Part of the sip:URI in the first line of the INVITE message. In CAS gateways (using the FXO Loop_Start_CAS protocol), the Called number field is empty. By setting the ReplaceEmptyDSTWithPortNumber parameter to 1, it is possible to set the channel preconfigured number as a called number.
2. The Redirect number and Redirect reason received in the SMDI message are mapped to the SIP Diversion header as a Tel URI or as a SIP URI (configurable using UseSIPURIForDiversionHeader). The Calling (source) number that is (optionally) received in the SMDI message is mapped to the SIP From header.
3. Manipulation rules can optionally be applied to the Called and Calling numbers.
4. For CAS gateways, usually the FXOLoopStart table is used. The table defines the correct signaling with PBX T1 CAS trunks. Different tables may be used per PBX. The CAS tables are different for SMDI and DTMF coding.

2.4.1 Relevant Parameters

The VM parameters can also be configured through the Web interface (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **DTMF & Supplementary** > **Supplementary Services Settings**).

- **VoiceMailInterface = 2:** Enables VM on the gateway and determines the communication method (i.e., **SMDI**) used between the PBX and the gateway:
- **SMDI = 1:** Enables SMDI on the gateway according to the BellCore standard
- **SMDI = 2:** Enables SMDI according to Ericsson MD-110 PBX
- **SMDI = 3:** Enables SMDI according to NEC NEAX 2400 PBX
- **SMDIlineIDLen:** Defines the length of the line identification field:
 - **Bellcore SMDI:** should be set to 7 (default).
 - **Ericsson MD-110 SMDI:** should be set between 2 to 5 according to the PBX configuration.
 - **NEC NEAX 2400:** typically, set to 6. For some PBX variants, the parameter can be set to 9.
- **SMDIInternalNumberLen:** Defines the PBX number length:
 - **Bellcore SMDI:** this parameter is not applicable.
 - **Ericsson MD-110 SMDI:** should be set to 2 - 10 (default = 5).
 - **NEC NEAX 2400:** typically, set to 6. For some PBX variants, the parameter can be set to 9.
- **SMDITimeOut:** Determines the time (in msec) that the gateway waits for an SMDI Call Status message before or after a Setup message is received. This parameter is used to synchronize the SMDI and Tel interfaces. If the timeout expires and only an SMDI message is received, the SMDI message is dropped. If the timeout expires and only a Setup message is received, the call is established. The valid range is 0 to 10,000 (i.e., 10 seconds). The default value is 2,000.
- **SMDIMWIMinInterval:** Defines the minimum time interval (in milliseconds) between sending subsequent MWI messages over SMDI. If 0, send as soon as possible. The default value is 250.
- **SMDIMWIQueueSize:** Queue size (in entries) for the above feature. The default value is 100.
- The following parameters must be defined for FXO gateways (not required for CAS):

- **TargetOfChannel0 = 100,2:** Enables delayed automatic dialing using HotDialToneDuration delay. The "100" (or other string) is the called number. This parameter must be repeated for all FXO gateway channels (TargetOfChannelX, X = 0 to 7).
- **HotDialToneDuration = 1:** Delay of one second between seizing the line and processing of SMDI message to overcome glare conditions.
- **TimeBetweenDigits = 1:** If DTMF digits are received during play of the Hot Dial tone, the digit collection is terminated after one second and the gateway starts to process the received SMDI messages.
- **EnableCallerID = 0:** Caller ID must be disabled for FXO gateways.
- **UseSIPURIForDiversionHeader:** Defines the URI format in the Diversion header.
 - 0 = 'tel:' (default), for example, 'tel:0000066242'
 - 1 = 'sip:', for example, 'sip:0000066242@10.2.3.4'
- **EnableVMURI:** If enabled, the INVITE includes "target" and "cause" parameters according to RFC 4458 (Voicemail URI).
- Tel to IP routing and manipulation parameters.
- TrunkGroup and TrunkGroupSettings: assign local numbers to endpoints that are matched to the SMDI Message Line Identifier field. For example, TrunkGroup = 1-8, 0010001.
- **SerialBaudRate:** Determines the RS-232 baud rate value. The valid range is any value.
- **SerialData:** Determines the RS-232 data bit value:
 - 7 = 7-bit
 - 8 = 8-bit (default)
- **SerialParity:** Determines the RS-232 parity value:
 - 0 = None (default)
 - 1 = Odd
 - 2 = Even
- **SerialStop:** Determines the RS-232 stop bit value:
 - 1 = 1-bit (default)
 - 2 = 2-bit
- **SerialFlowControl:** Determines the RS-232 flow control value:
 - 0 = None (default)
 - 1 = Hardware
- **ECNLPMODE = 1:** disables Echo Canceller NLP. This can improve the gateway's detection of DTMF digits during the playing of IVR announcements.
- **EchoCancellerAggressiveNLP=0.**

2.4.2 Examples

Example of a Received SMDI BellCore Message:

```
MD0010003N0000066242 0000061382
MessageDeskNum:001 PositionNum:0003 CallType:N
ForwardDN:0000066242 CallingDN:0000061382 CallingDNStatus:
CallingName:
```

Example of an INVITE Message with Diversion Header:

```
INVITE sip:0010003@157.55.89.37;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacZjvZlHl;alias
Max-Forwards: 70
From: <sip:0000061382@10.197.96.30>;tag=1c2794514316
To: <sip:0010003@157.55.89.37;user=phone>
Call-ID: 2415530342UgsW@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:0000061382@10.197.96.30;transport=tcp>
Diversion: <tel:0000066242>;reason="no-answer";screen=no
;privacy=off
Content-Type: application/sdp
Content-Length: 246

v=0
o=AudiocodesGW 296442 595422 IN IP4 10.197.96.30
s=Phone-Call
c=IN IP4 10.197.96.30
t=0 0
m=audio 6020 RTP/AVP 0 8 96
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
a=sendrecv
```

Example of an INVITE Message with voicemail URI and Diversion Header:

```
INVITE sip:voicemail@157.55.89.37; target=sip:
0000066242%40audiocodes.com;cause=408;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacZjvZlHl;alias
Max-Forwards: 70
From: <sip:0000061382@10.197.96.30>;tag=1c2794514316
To: <sip:0010003@157.55.89.37;user=phone>
Call-ID: 2415530342UgsW@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:0000061382@10.197.96.30;transport=tcp>
Diversion: <tel:0000066242>;reason="no-
answer";screen=no;privacy=off
```

```
Content-Type: application/sdp
Content-Length: 246
...
```

SIP Diversion header examples:

- Diversion: <tel:3222>;reason="no-answer"
- Diversion: <tel:555>;reason="unconditional"
- Diversion: <tel:444>;reason="user-busy"
- Diversion: <sip:1234@gatewayname>;reason="no-answer"

2.5 Transfer Request from the IP Voice Mail to PBX

After a call from the PBX to the VM is established (as described in Section 2.4), the VM may resolve to transfer the call to an operator or to another PBX extension (or even to an external number).

The Transfer request is sent by the IP VM using a REFER message. If the IP address in the Refer-To URI is the address of the gateway itself, the gateway initiates a call transfer to the PBX extension number, and then releases the line after the call transfer is complete. If the IP address in the Refer-To URI is different from the address of the gateway, the gateway establishes a new SIP connection to that address. (**Note:** If FQDN is used in the Refer-To header instead of the gateway numerical IP address, you should configure in the gateway's Internal DNS table the association between this FQDN and 127.0.0.1.)

The FXO/CAS gateways support three types of transfers: Blind Transfer, Semi-Supervised, and Supervised Transfer.

The FXO / CAS (normal) gateways perform a Blind Transfer using the following sequence:

```
Flash-hook (Wink for CAS) -> Wait -> Dial destination -> Wait ->
Release line
```

The CAS NFA gateway performs a 'Blind Transfer' using the following sequence:

```
Wink -> Wait for detection of confirmation Wink -> Dial
destination -> Wait -> Release line
```

The FXO/CAS (normal) gateways perform a Semi-Supervised Transfer using the following sequence:

```
Flash-hook (Wink for CAS) -> Wait -> Dial destination -> Wait
for Busy/Reorder -> Release line to complete the call transfer
if Busy or Reorder tones are not detected.
If the gateway detects Busy or Reorder tones during
WaitForBusyTime timeout, the gateway sends a SIP NOTIFY message
with a failure reason in the NOTIFY body and generates
additional flash-hook (Wink for CAS) to restore the original
call. The transfer operation is not completed in this scenario.
```

2.5.1 Relevant Parameters

The relevant parameters are described as follows:

- **EnableTransfer = 1:** must be set to 1 to enable transfer.
- **LineTransferMode:**
 - **1 (Blind Transfer):** the FXO gateway activates Blind Transfer to the PBX using the already seized FXO line. The FXO gateway performs Blind Transfer by generating a hook-flash, waiting for WaitForDialTime, dialing Refer-To number, and then finally going on-hook to complete the call transfer.
 - **2 (Semi-Supervised Transfer):** the FXO gateway activates Semi-Supervised Transfer to the PBX using the already seized FXO line. The FXO gateway generates a hook-flash, waits for WaitForDialTime, and dials the Refer-To number. If during WaitForBusyTime, no Busy or Reorder tones are detected, the gateway completes the call transfer by releasing the line. If during WaitForBusyTime, the gateway detects Busy or Reorder tones, the transfer operation is cancelled, the gateway sends a SIP NOTIFY message with a failure reason in the NOTIFY body (such as 486 if busy tone was detected), and generates an additional hook-flash towards the FXO line to restore connection to the original call.
 - **3 (Supervised Transfer):** the FXO gateway activates Supervised Transfer to the PBX using the already seized FXO line. The FXO gateway generates a hook-flash, waits for WaitForDialTime, and dials the Refer-To number. The gateway completes the call transfer by going on-hook only after detection of the transferred party answer. The answer supervision is achieved by voice detection, and requires the setting of EnableVoiceDetection to 1. If a busy / reorder tone is detected before answer, the call behaves in the same manner as for 2 (Semi-Supervised Transfer), and the transfer is cancelled.



Note: The LineTransferMode parameter also affects the transfer method of the CAS digital gateway (see the description below for the TrunkTransferMode_x parameter).

- **TrunkTransferMode_x = 3 (CAS Normal) or 1 (CAS NFA) (digital gateways only)**
 When a SIP REFER message is received, the CAS gateway performs a Blind Transfer (If LineTransferMode = 1) by executing a CAS Wink, waiting for WaitForDialTime, dialing the Refer-To number, and then releasing the call.
 If LineTransferMode = 2, the gateway performs a Semi-Supervised Transfer. The gateway executes a CAS Wink, waits for WaitForDialTime, and dials the Refer-To number. If during WaitForBusyTime no Busy or Reorder tones are detected, the gateway completes the call transfer by releasing the line. If during WaitForBusyTime, the gateway detects Busy or Reorder tones, the transfer operation is cancelled, the gateway sends a SIP NOTIFY message with a failure reason in the NOTIFY body (such as 486 if busy tone was detected), and generates an additional Wink towards the CAS line to restore connection to the original call.

If LineTransferMode = 3, the CAS gateway performs Supervised Transfer to the PBX. The gateway executes a CAS Wink, waits for WaitForDialTime, and dials the Refer-To number. The gateway completes the call transfer by releasing the line only after detection of the transferred party answer. The answer supervision is achieved by voice detection, and requires the setting of EnableVoiceDetection to 1, EnabledDSIPMDetectors to 1, and feature key containing IPMDetector DSP option.

**Notes:**

- A specific CAS table supporting wink is required.
- Blind or Semi-Supervised Transfer is performed without waiting for an answer from the remote party.
- Supervised Transfer is performed only after the remote party answers the call and its voice is detected.

- **WaitForDialTime:** determines the delay between the time the line is seized and the dialing begins during the establishment of an IP→Tel call for FXO and CAS gateways. The valid range (in milliseconds) is 0 to 20,000 (i.e., 20 seconds). The default value is 1,000 (i.e., 1 second).
- **WaitForBusyTime:** applicable for Semi-Supervised Transfer (if LineTransferMode = 2). It determines the time for detection of Busy or Reorder tones before releasing the line to complete the call transfer. If during this time, Busy or Reorder tones are detected, the transfer operation is cancelled. The valid range (in milliseconds) is 0 to 10,000 (i.e., 10 seconds). The default value is 2,000 (i.e., 2 seconds).

Additional related parameters:

- **FlashHookPeriod (FXO gateways only):** determines the hook-flash generation period (in msec). The valid range is 300 to 1,500 (default is 400).



Note: For FXO gateways, a constant of 90 msec must be added to the required hook-flash period. For example, to generate a 450 msec hook-flash, set FlashHookPeriod to 540.

- **DTMF dialing parameters:** DTMFDigitLength, DTMFInterDigitInterval, DTMFVolume.
- **XferPrefixIP2Tel:** Defines the prefix that is added to the destination number received in the SIP Refer-to header (in IP-to-Tel calls).

2.6 MWI Notification from IP Voice Mail to PBX

The IP VM uses SIP NOTIFY messages to inform the gateway of Message Waiting Indications (MWI). The gateway may subscribe to an MWI server (a single subscription per gateway) using the SIP MWI package (RFC 3842 Message Summary and Message Waiting Indication Event Package for SIP).

After a successful subscription, the MWI server can start sending NOTIFY messages to the gateway with message waiting indications. The extension number sent to the PBX is provided in the Message Account header in a NOTIFY message, as shown in the example below using extension number 401:

```
Messages-Waiting: yes
Message-Account: sip:401@10.13.77.10
```

After being informed by the VM of an MWI, the gateway sends an Operate MWI (Turn On) message over the serial connection to the PBX. In this message, the gateway includes the extension number received in the NOTIFY message ('401' in the example above). A Remove MWI (Turn Off) message is used to remove the MWI.



Notes:

- The gateway also accepts MWI Notify messages without prior subscription to the MWI server.
- If the Message-Account parameter is missing, the gateway uses the SIP URI User Part from the To header, as the Account number.
- The IP to Tel Routing table can be used to route the MWI messages to a specific Trunk Group, according to URI User Part in the From header of the received NOTIFY message.

2.6.1 Relevant Parameters

The relevant parameters are described as follows:

- VoiceMailInterface = 2 (for SMDI)
- SubscriptionMode = 1 (subscription per gateway): determines the method the gateway uses to subscribe to an MWI server.
- SubscriptionMode = 0 (subscription per endpoint). For example, all MP-118 FXS/FXO endpoints will be subscribed.
- Username = Gateway: account name used for subscription to the MWI server.
- EnableMWI = 1
- EnableMWISubscription = 1 (optional)
- MWIExpirationTime = 7200 (seconds)
- SubscribeRetryTime = 120 (seconds)
- MWIServerIP = xxx.xxx.xxx.xxx: IP address of the MWI server.
- MWIServerTransportType: -1 for not configured (default); 0 for UDP; 1 for TCP; 2 for TLS. If not configured (-1), use the SIP Transport according to the SIPTransportType parameter.
- SMDI = 1, 2 or 3
- ECNLPMMode = 1: disables the Echo Canceller NLP. This can improve the gateway's detection of DTMF digits during the playing of IVR announcements.

- EchoCancellerAgressiveNLP=0.
- Additional related parameters:
 - SerialBaudRate
 - SerialData
 - SerialParity
 - SerialStop
 - SerialFlowControl

2.6.2 Examples

Subscribe Example:

```

SUBSCRIBE sip:Gateway@10.33.2.10 SIP/2.0
Via: SIP/2.0/UDP 10.33.4.103;branch=z9hG4bKacAqDJNSC
Max-Forwards: 70
From: <sip: Gateway @10.33.4.103>;tag=1c133925930
To: <sip:Gateway@10.33.4.103>
Call-ID: 2658731555ISYm@10.33.4.103
CSeq: 1 SUBSCRIBE
Contact: <sip:Gateway@10.33.4.103>;expires=7200
Event: message-summary
Supported: em,timer,replaces,path
Allow: REGISTER, OPTIONS, INVITE, ACK, CANCEL, BYE, NOTIFY,
PRACK, REFER, INFO, SUBSCRIBE, UPDATE
Expires: 7200
User-Agent: Audiocodes-Sip-Gateway-MP-104 FXO/v.4.40.0.200
Content-Length: 0
  
```

Notify Example:

```

NOTIFY sip:Gateway@10.33.2.10 SIP/2.0
Via: SIP/2.0/UDP 10.33.2.10;branch=z9hG4bKadfe4ko
To: <sip:Gateway@10.33.4.103>;tag=1c133925930
From: <sip:Gateway@10.33.4.103>;tag=1111
Call-ID: 2658731555ISYm@10.33.4.103
CSeq: 1 NOTIFY
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Event: message-summary
Content-Type: application/simple-message-summary
Content-Length: 100

Messages-Waiting: yes
Message-Account: sip:401@10.33.4.103
  
```


2.6.3 Interworking Bellcore SMDI "INV" and "BLK" to SIP NOTIFY

The gateway supports the interworking of Bellcore SMDI "INV" and "BLK" error messages to SIP NOTIFY messages. The "INV" message indicates that the subscriber directory number (DN) provided in the SMDI MWI request is invalid. The "BLK" message indicates that the PBX is currently unable to execute the message desk request (MSR).

The IP VM can implement a keep-alive mechanism to verify that the SMDI link with the PBX is ok. The IP VM does this by periodically sending a SIP MWI NOTIFY message with an invalid account number to the PBX (via the gateway). As a result of the invalid account, the PBX replies with an SMDI "INV" or "BLK" error message. Upon receipt of this error message, the gateway sends a SIP NOTIFY message with the "Message-status: failure INV (or BLK)" field to the IP VM, as shown in the example below:

```
NOTIFY sip:abc@domain.com:5060 SIP/2.0
Via: SIP/2.0/UDP 10.55.18.148:5061;branch=z9hG4bK-15549-27-0
To: <sip:abc@domain.com:5060>;tag=1c281018370
From: <sip:Gateway@gw3.domain.com:5060>
Call-ID: 27-15549@10.55.18.148
CSeq: 1 NOTIFY
Max-Forwards: 70
Contact: <sip:gw3@10.55.18.148:5060>
Event: message-summary
Content-Type: application/simple-message-summary
Content-Length: 48

Message-account: 1212
Message-status: failure INV (or BLK)
```

The NOTIFY messages are sent to the Proxy Set IP address as defined for the NotificationIPGroup.

2.7 PBX Disconnect Supervision

This section describes PBX disconnect supervision.

2.7.1 General Disconnect Methods

The gateway supports the following methods for detecting PBX line disconnection:

- Detection of DTMF disconnect code (supported by certain PBXs).
- CAS signaling (such as receive of AB=1x from PBX, for loop start CAS signalling).
- Detection of Busy or Reorder tones (both applicable for CAS and FXO gateways).
- Detection of Current Disconnect for FXO gateways (EnableCurrentDisconnect = 1).
- Detection of Polarity Reversal for FXO (EnableReversalPolarity = 1).
- Silence detection for preconfigured timeout. The CAS gateway supports silence detection only if EnableDSPIPMDetectors = 1.
- Max Call duration.

It's possible to simultaneously configure several disconnect supervision methods.

2.7.2 DTMF Disconnect Code

This is a DTMF code sent by the PBX to the gateway notifying that the VM call has been terminated. When the gateway identifies that a call is terminated by the PBX, it closes the IP session and releases the line towards the PBX.

A digit pattern received from the Tel side notifies the gateway to disconnect the call.

Relevant parameter: TelDisconnectCode.

For example:

- **Panasonic:** #9; TelDisconnectCode = #9 (for Panasonic PBX)
- **Toshiba:** D; TelDisconnectCode = D (for Toshiba PBX)

2.7.3 Relevant Parameters

The relevant parameters are described as follows:

- DisconnectOnBusyTone = 1
- EnableCurrentDisconnect = 1
- CurrentDisconnectDuration = 500
- EnableReversalPolarity = 1
- EnableSilenceDisconnect = 1
- FarEndDisconnectSilencePeriod = 60 (for 60 seconds of silence)
- MaxCallDuration = 120 (seconds)
- TelDisconnectCode = '#9'
- EnableDSPIPMDetectors = 1 (**Note:** Applicable for CAS gateways and should be enabled in the gateway Feature Key. The number of gateway channels may be reduced -- please refer to the User's Manual.)

2.8 Calls from Voice Mail to PBX via Analog FXO Gateway

When the VM initiates a call to the PBX via the gateway, the gateway receives an INVITE message and uses the User Part of the request URI as a called number. The FXO gateway

seizes one of its available lines, waits for dial tone detection, and then dials the DTMF digits. There is also an option to configure the gateway to dial without waiting for a dial tone (`IsWaitForDialTone = 0`), instead it waits for `WaitForDialTime` before dialing the digits. The gateway sends a 180 response immediately after receiving the INVITE message.

The subsequent gateway behavior depends on whether or not the Answer Supervision feature is enabled. If Answer Supervision is enabled, the gateway detects when the remote party answers the call.

Currently, there are four answer supervision methods:

- Detection of Polarity Reversal signal sent by the PBX to the gateway over the analog line
- Speech Detection of the answering party (if the party says “hello” or any other word)
- Detection of fax or modem answer tone
- DTMF code sent by the PBX

If Answer Supervision is not used (`EnableReversalPolarity = 0`, `EnableVoiceDetection = 0`, and no DTMF Connect code), the FXO gateway sends a 200 OK response immediately after completing the dialing. If Answer Supervision is used, a 200 OK + SDP are later sent only if the remote party answers the call.

2.8.1 Relevant Parameters

The relevant parameters are described as follows:

- `IsTwoStageDial = 0`
- `EnableVoiceDetection = 1`
- `TELConnectCode = 'xxx'`: defines the DTMF sequence sent by the PBX if the remote party answers the call. This is supported by some PBXs.
- `ChannelSelectMode = 1`: activates the 'Cyclic ascending' channel selection mode.
- Additional related parameters:
 - `EnableEarlyMedia`
 - `EnableReversalPolarity`
 - `IsWaitForDialTone`
 - `WaitForDialTime`
 - DTMF dialing parameters: `DTMFDigitLength`, `DTMFInterDigitInterval`, `DTMFVolume`
 - `TrunkGroup` and `TrunkGroupSettings`: assign local numbers to gateway endpoints and define port allocation method per trunk group.

2.9 Calls from Voice Mail to PBX via Digital CAS Gateway

The T1 Loop Start access lines are used for connection between the PBX and Voice Mail (or gateway). The PBX typically uses the Foreign Exchange Office (FXO) interface type while the VM/gateway uses the Foreign Exchange Station (FXS) interface type. The disadvantage of loop-start signaling is the inability to be notified when a remote party answers the VM-to-PBX call (answer supervision). Some PBXs notify the VM that the call is answered by sending a special DTMF code.

Currently, there are three answer supervision methods:

- Speech Detection of the answering party (if the party says “hello” or any other word)
- Detection of fax or modem answer tone
- DTMF code sent by the PBX

When the VM initiates a call to the PBX via the gateway, the gateway receives an INVITE message and uses the User Part of the request URI as a called number. The gateway seizes one of its available T1 lines, waits for dial tone detection, and then dials the DTMF digits. There is also an option to configure the gateway to dial without waiting for the dial tone (by using a different CAS table). If EarlyMedia feature is enabled, the CAS gateway sends a 183 + SDP response immediately after completing the dialing. If EarlyMedia is disabled, the gateway sends instead a 180 Ringing response.

The subsequent gateway behavior depends on the used CAS table and whether or not the Speech Detection feature is enabled. If EnableVoiceDetection is set to 1 and EnableDSPIPMDetectors is set to 1, the gateway detects the answering party (if the party says “hello” or any other word), and then sends 200 OK + SDP. If voice detection is disabled, 200 OK + SDP is sent only after the timeout as defined in the CAS table (default is 20 sec).



Notes:

- To enable voice detection, the gateway Feature Key should contain the “IPMDetector” DSP option.
- Setting EnableDSPIPMDetectors to 1 can reduce the number of gateway channels (refer to the User’s Manual).

2.9.1 Relevant Parameters

The relevant parameters are described as follows:

- EnableVoiceDetection = 1
- EnableDSPIPMDetectors = 1
- WaitForDialTime = 2000
- TELConnectCode = ‘xxx’: defines the DTMF sequence sent by the PBX if the remote party answers the call. This is supported by some PBXs.
- Additional related parameters:
 - EnableEarlyMedia
 - DTMF dialing parameters: DTMFDigitLength, DTMFInterDigitInterval, DTMFVolume
 - TrunkGroup and TrunkGroupSettings: assign local numbers to gateway endpoints and define port allocation method per trunk group.

3 DTMF Operation

This section describes DTMF operation.

3.1 Calls from PBX to Voice Mail

When the gateway detects a ringing signal, it seizes the line and collects the received DTMF digits. The collected digits are compared against pre-configured patterns to extract the Redirect Number (usually identifies the VM box number), Redirect Reason (forward reason) and, if available, the Source Number. The patterns are configured per PBX according to its specifications.



Note: For some PBXs, if a direct call is made to the Voice Mail, no DTMF digits are dialed by the PBX. In such a scenario, the gateway, after two seconds, sends an INVITE message to the VM with a preconfigured number as a User Part of the sip:URI.

The gateway sends an INVITE message to the IP VM. If a Redirect Number is detected from the PBX, the INVITE message includes a Diversion header that contains the Redirect Number and Redirect Reason (according to the IETF draft draft-levi-sip-diversion-08). This call is regarded as a 'forwarded call' (to the VM from the 'Redirect' extension) due to a Busy, No Answer, or Do Not Disturb (DND) condition (or some other unknown reason). It is also possible to configure the gateway to include "target" and "cause" parameters in the INVITE message according to RFC 4458 (Voicemail URI). This option is enabled by setting the *ini* file parameter EnableVMURI to 1.

The From header of the same SIP INVITE contains the extracted Source Number (if exists as part of the received digits pattern) as its User Part. If a Source Number isn't received, the From header only contains the gateway's host name (unless a manipulation rule is used to add a calling number). All INVITE messages generated by the gateway contain the original dialed number as the User Part of the SIP URI Request. Sometimes, this "Called number" should be modified or removed using number manipulation rules.

To summarize, a call from a PBX to a VM is processed as follows:

1. The Called Number is collected using the regular methods:
 - **FXO:** by using interdigit timeout or digit-mapping rules. In addition, it's possible to use delayed Automatic Dialing to a preconfigured number (using the parameter HotlineToneDuration) if no DTMF digits are received from the PBX. This Called Number appears as the User Part of the sip:URI in the first line of the INVITE message.
 - **CAS:** for FXO loop-start protocol, the Called Number is collected using digit map rules. Therefore, it's mandatory to configure the digitmap parameter for FXO loop-start CAS protocol. In other CAS protocols, the Called Number is collected using AudioCodes' CAS table.
2. The Redirect number and Redirect reason extracted from the received Called Number are mapped to the SIP Diversion header as a Tel URI or as a SIP URI (configurable using UseSIPURIForDiversionHeader). The Calling (source) number that is (optionally) extracted from the received Called Number is mapped to the SIP From header.
3. Manipulation rules can optionally be applied to the Called and Calling numbers.
4. For CAS gateways, usually the FXOLoopStart table is used. The table defines the proper signaling with PBX T1 CAS trunks. Different tables may be used for each PBX. The CAS tables are different for SMDI and DTMF coding.

3.1.1 Pattern Syntax

The digit patterns generated by the PBX are composed of the following notations:

Table 3-1: PBX Digit Pattern Notations

Digit Pattern Notation	Description
0-9, A-D, #, *	Specific DTMF digits.
R	A single Redirect Number digit.
S	A single Source Number digit.
X	Any digit.
'd' (single dot followed by a specific digit)	Repeat the last rule until digit 'd' is received (digit 'd' is ignored).
'd.' (single specific digit followed by a dot)	Match and ignore multiple identical 'd' digits. This rule can be used to skip multiple identical digits such as '*****'.

For example:

- **#03RRR#S.#** : collects a three-digit Redirect number, and Source number of variable size, until a '#' sign is received.
- **#01#R.#** : collects digits to Redirect Number until the digit '#' is received.
- **X.*RRR**: ignores all digits until the digit '*' is received ('*' is also ignored), and collects a three-digit Redirect number.
- ***.S.*.R*.**: collects source (1234) and redirect (567) numbers from the following string *****1234**567*****



Note: The forwarded patterns must contain the Redirect Number ('R') and may also contain the Source Number ('S'). The internal / external call patterns may contain the Source Number ('S'), but not the Redirect Number.

The general-forward-pattern parameter DigitPatternForwardNoReason can be used to indicate the forward reason when it isn't specified by the PBX or when it isn't recognized by the gateway.

Several PBXs use an IVR dialog with the caller to locate the mailbox number instead of directly communicating with the IP VM using digit patterns. Therefore, if after a short timeout, no digits are received from the PBX, an INVITE message is sent to the IP VM with a preconfigured number. This is performed by using the Hotline Dialing feature (for analog gateways).

Relevant parameters: TargetOfChannelX; HotLineDialToneDuration.

3.1.2 Relevant Parameters

The VM parameters can also be configured through the Web interface (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **DTMF & Supplementary** > **Supplementary Services Settings**).

- **VoiceMailInterface = 1:** enables VM on the gateway and determines the communication method used between the PBX and the gateway:
 - 0 = None (default)
 - 1 = DTMF
 - 2 = SMDI
- **DigitPatternForwardOnBusy** and **DigitPatternForwardOnBusyExt:** determines the digit pattern used by the PBX to indicate a Call Forward on Busy. The valid range is a 120-character string. It's possible to define two digit patterns for Call Forward On Busy reason. This is for PBXs that use two DTMF patterns to describe such VM calls. A typical example is when different patterns are used for internal and external calls.
- **DigitPatternForwardOnNoAnswer** and **DigitPatternForwardOnNoAnswerExt:** determines the digit pattern used by the PBX to indicate Call Forward on No Answer. The valid range is a 120-character string.
- **DigitPatternForwardOnDND** and **DigitPatternForwardOnDNDExt:** determines the digit pattern used by the PBX to indicate Call Forward on Do Not Disturb. The valid range is a 120-character string.
- **DigitPatternForwardNoReason** and **DigitPatternForwardNoReasonExt:** determines the digit pattern used by the PBX to indicate Call Forward with No Reason. The valid range is a 120-character string.
- **DigitPatternInternalCall:** determines the digit pattern used by the PBX to indicate an internal call. The valid range is a 120-character string.
- **DigitPatternExternalCall:** determines the digit pattern used by the PBX to indicate an external call. The valid range is a 120-character string.
- **TelDisconnectCode:** determines a digit pattern that when received from the Tel side, indicates the gateway to disconnect the call. The valid range is a 25-character string.
- The following parameters must be defined for FXO gateways (not required for CAS):
 - **TargetOfChannel0 = 100,2:** enables delayed automatic dialing using HotDialToneDuration delay. The '100' (or other string) is the called number that is sent as a User Part in INVITE URI if no DTMF digits are detected during the HotDialToneDuration delay. This parameter must be repeated for all FXO gateway channels (TargetOgChannelX, X=0 to 7).
 - **HotDialToneDuration = 3:** delay of three seconds between seizing the line and performing the automatic dialing if no DTMF digits are received from the PBX. During these three seconds, the FXO plays a dial tone to the PBX. The dial tone can be suppressed by using a modified Call Progress Tone file containing a "muted" dial tone.
 - **EnableCallerID = 0:** Caller ID must be disabled for FXO gateways.
 - **IsSpecialDigits = 1:** must be set to 1, since most PBXs use * and / or # in their digit patterns.
- **ECNLPMODE = 1:** Disables Echo Canceller NLP. This can improve the gateway's detection of DTMF digits during the playing of IVR announcements.

- **EchoCancellerAgressiveNLP=0.**
- Additional related parameters:
 - **DigitMapping:** must be set to collect PBX digits (recommended value: [x*#][x*#.t]).
 - **TimeBetweenDigits:** Interdigit timeout.
 - **MaxDigits:** must be large enough to capture all PBX codes (recommended value is 30).
 - **UseSIPURIForDiversionHeader:** sets the URI format in the Diversion header.
 - ◆ 0 = 'tel:' (default), for example, 'tel:0000066242'
 - ◆ 1 = 'sip:', for example, 'sip:0000066242@10.2.3.4'
 - **EnableVMURI:** if enabled, the INVITE includes "target" and "cause" parameters according to RFC 4458 (Voicemail URI).
 - Tel to IP routing and manipulation parameters.
 - **TrunkGroup and TrunkGroupSettings:** assign local numbers to gateway endpoints.

3.1.3 Examples

Pattern examples:

- **Avaya:**
 - DigitPatternForwardNoReason = '#02#S.#R.#'
 - DigitPatternInternalCall = '#00#S.##'
- **NEC, Nortel:** doesn't have separate reasons for Call Forward. They always dial the Redirect number only.
 - DigitPatternForwardNoReason = R.

Example of an INVITE Message with Diversion Header:

```
INVITE sip:4105@157.55.89.37;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacZjvZlH1;alias
Max-Forwards: 70
From: <sip:1000@10.197.96.30>;tag=1c2794514316
To: <sip:4105@157.55.89.37;user=phone>
Call-ID: 2415530342UgsW@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:1000@10.197.96.30;transport=tcp>
Supported: em,100rel,timer,replaces,path Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,S
UBSCRIBE,UPDATE
Diversion: <tel:4105>;reason="no-answer";screen=no;privacy=off
Content-Type: application/sdp
Content-Length: 246

v=0
o=AudiocodesGW 296442 595422 IN IP4 10.197.96.30
s=Phone-Call
c=IN IP4 10.197.96.30
t=0 0
m=audio 6020 RTP/AVP 0 8 96
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
```



```
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
a=sendrecv
```

Example of an INVITE Message with voicemail URI and Diversion Header:

```
INVITE sip:voicemail@157.55.89.37; target=sip:
4105%40audiocodes.com;cause=408;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacZjvZlH1;alias
Max-Forwards: 70
From: <sip:1000@10.197.96.30>;tag=1c2794514316
To: <sip:voicemail@157.55.89.37;user=phone>
Call-ID: 2415530342UgsW@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:1000@10.197.96.30;transport=tcp>
Diversion: <tel:4105>;reason="no-answer";screen=no;privacy=off
Content-Type: application/sdp
Content-Length: 246
...
```

SIP Diversion header examples:

- Diversion: <tel:3222>;reason="no-answer"
- Diversion: <tel:555>;reason="unconditional"
- Diversion: <tel:444>;reason="user-busy";
- Diversion: <sip:1234@gatewayname>;reason="no-answer"

3.2 Transfer Request from IP Voice Mail to PBX

After a call from the PBX to the VM is established (as described in Section 3.1), the VM may resolve to transfer the call to an operator or to another PBX extension (or even to an external number).

The Transfer request is sent by the IP VM using a REFER message. If the IP address in the Refer-To URI is the address of the gateway itself, the gateway initiates a call transfer to the PBX extension number, and then releases the line after the call transfer is complete. If the IP address in the Refer-To URI is different from the address of the gateway, the gateway establishes a new SIP connection to that address. (**Note:** If FQDN is used in the Refer-To header instead of in the gateway numerical IP address, you should configure in the gateway's Internal DNS table the association between this FQDN and 127.0.0.1.)

The FXO/CAS gateways support three types of transfers: Blind Transfer, Semi-Supervised and Supervised Transfer.

The FXO / CAS (normal) gateways perform a Blind Transfer using the following sequence:

```
Flash-hook (Wink for CAS) -> Wait -> Dial destination -> Wait ->
Release line
```

The CAS NFA gateway performs a 'Blind Transfer' using the following sequence:

```
Wink -> Wait for detection of confirmation Wink -> Dial
destination -> Wait -> Release line
```

The FXO/CAS (normal) gateways perform a Semi-Supervised Transfer using the following sequence:

```
Flash-hook (Wink for CAS) -> Wait -> Dial destination -> Wait
for Busy/Reorder -> Release line to complete the call transfer
if Busy or Reorder tones are not detected.
If the gateway detects Busy or Reorder tones during
WaitForBusyTime timeout, the gateway sends a SIP NOTIFY message
with a failure reason in the NOTIFY body and generates
additional flash-hook (Wink for CAS) to restore the original
call. The transfer operation is not completed in this scenario.
```

3.2.1 Relevant Parameters

The relevant parameters are described as follows:

- **EnableTransfer = 1:** must be set to 1 to enable transfer.
- **LineTransferMode:**
 - **1 (for Blind Transfer):** the FXO gateway activates Blind Transfer to the PBX using the already seized FXO line. The FXO gateway performs Blind Transfer by generating a hook-flash, waiting for WaitForDialTime, dialing Refer-To number, and then finally going on-hook to complete the call transfer.
 - **2 (for Semi-Supervised Transfer):** the FXO gateway activates Semi-Supervised Transfer to the PBX using the already seized FXO line. The FXO gateway generates a hook-flash, waits for WaitForDialTime, and dials the Refer-To number. If during WaitForBusyTime, no Busy or Reorder tones are detected, the gateway completes the call transfer by releasing the line. If during WaitForBusyTime, the gateway detects Busy or Reorder tones, the transfer operation is cancelled, the gateway sends a SIP NOTIFY message with a failure reason in the NOTIFY body (such as 486 if busy tone was detected), and generates an additional hook-flash towards the FXO line to restore connection to the original call.
 - **3 (for Supervised Transfer):** the FXO gateway activates Supervised Transfer to the PBX using the already seized FXO line. The FXO gateway generates a hook-flash, waits for WaitForDialTime, and dials the Refer-To number. The gateway completes the call transfer by going on-hook, only after detection of the transferred party answer. The answer supervision is achieved by voice detection, and requires setting of EnableVoiceDetection to 1.
If a busy / reorder tone is detected before answer, the call behaves in the same manner as for 2 (Semi-Supervised Transfer), and the transfer is cancelled.



Note: The LineTransferMode parameter also affects the transfer method of the CAS digital gateway (see the description below for the TrunkTransferMode_x parameter).

- **TrunkTransferMode_x = 3 (CAS Normal) or 1 (CAS NFA) (digital gateways only)**
When a SIP REFER message is received, the CAS gateway performs a Blind Transfer (If LineTransferMode = 1) by executing a CAS Wink, waiting for WaitForDialTime, dialing the Refer-To number, and then releasing the call.
 - If LineTransferMode = 2, the gateway performs a Semi-Supervised Transfer. The gateway executes a CAS Wink, waits for WaitForDialTime, and dials the Refer-To number. If during WaitForBusyTime no Busy or Reorder tones were detected, the gateway completes the call transfer by releasing the line. If during WaitForBusyTime, the gateway detects Busy or Reorder tones, the transfer operation is cancelled, the gateway sends a SIP NOTIFY message with a failure reason in the NOTIFY body (such as 486 if busy tone was detected), and generates an additional Wink towards the CAS line, to restore connection to the original call.
 - If LineTransferMode = 3, the CAS gateway performs Supervised Transfer to the PBX. The gateway executes a CAS Wink, waits for WaitForDialTime, and dials the Refer-To number. The gateway completes the call transfer by releasing the line only after detection of the transferred party answer. The answer supervision is achieved by voice detection, and requires setting EnableVoiceDetection to 1, EnabledSPIPMDetectors to 1, and feature key containing IPMDetector DSP option.



Notes:

- A specific CAS table supporting wink is required.
- Blind or Semi-Supervised Transfer is performed without waiting for an answer from the remote party.
- Supervised Transfer is performed only after the remote party answers the call and its voice is detected.

- **WaitForDialTime:** determines the delay between the time the line is seized and the start of dialing during the establishment of an IP→Tel call for FXO and CAS gateways. The valid range (in milliseconds) is 0 to 20,000 (i.e., 20 seconds). The default value is 1,000 (i.e., 1 second).
- **WaitForBusyTime:** applicable for Semi-Supervised Transfer (if LineTransferMode = 2). It determines the time for detection of Busy or Reorder tones before releasing the line to complete the call transfer. If during this time, Busy or Reorder tones are detected, the transfer operation is cancelled. The valid range (in milliseconds) is 0 to 10,000 (i.e., 10 seconds). The default value is 2,000 (i.e., 2 seconds).
- Additional related parameters:
 - FlashHookPeriod (FXO gateways only): determines the hook-flash generation period (in msec). The valid range is 300 to 1,500 (default is 400).



Note: For FXO gateways, a constant of 90 msec must be added to the required hook-flash period. For example, to generate a 450 msec hook-flash, set FlashHookPeriod to 540.

- DTMF dialing parameters: DTMFDigitLength, DTMFInterDigitInterval, DTMFVolume
- **XferPrefixIP2Tel:** Defines the prefix that is added to the destination number received in the SIP Refer-to header (in IP-to-Tel calls).

3.3 MWI Notification from IP Voice Mail to PBX

The IP VM uses SIP NOTIFY messages to inform the gateway of Message Waiting Indications (MWI). The gateway may subscribe to an MWI server (a single subscription per gateway) using the SIP MWI package (RFC3842 Message Summary and Message Waiting Indication Event Package for SIP).

After a successful subscription, the server can start sending NOTIFY messages to the gateway with MWI. The extension number sent to the PBX is provided in the Message Account header in a NOTIFY message (shown in [Figure 3-1](#), extension number 401).

Figure 3-1: MWI Header Fields

```
Messages-Waiting: yes
Message-Account: sip:401@10.13.77.10
```

After being informed by the VM of MWI, the gateway notifies the PBX. It allocates a Trunk/Hunt Group according to the Source IP (MWIServerIP), Destination Number (Message-Account), and Source Number. The allocation procedure (IP to Trunk/Hunt group routing table) is similar to the mechanism used to allocate incoming IP-to-Tel calls (for detailed information, refer to the User's Manual).

The gateway seizes the line, and then dials a number to the PBX that is composed of a digit code and an extension number. The extension number is the number received from the VM in the NOTIFY message ('401' in the example above). The digit code is a configurable pattern that indicates if there are / or aren't any waiting messages.

The procedure performed by the gateway is illustrated below:

Seizes a free line → Waits (refer to Note 1) → Dials digit code and extension → Waits → Releases the line.

Notes:

- The FXO gateway waits for WaitForDialTime before dialing. The FXO doesn't detect a dial tone before dialing the MWI code. The CAS gateway can be configured (using CAS table), to detect a dial tone before dialing or start dialing, after a preconfigured delay, without detection of a dial tone.
- The gateway also accepts MWI Notify messages without Subscription to the MWI server.
- If the Message-Account parameter is missing, the gateway uses the SIP URI User Part from the To header, as the Account number.
- The IP to Tel Routing table can be used to route the MWI messages to a specific Trunk Group, according to URI User Part in the From header of the received NOTIFY message.



3.3.1 Relevant Parameters

The relevant parameters are described as follows:

- VoiceMailInterface = 1 (for DTMF)
- SubscriptionMode = 1 (subscription per gateway): determines the method the gateway uses to subscribe to an MWI server.
- SubscriptionMode = 0 (subscription per endpoints). For example, all MP118 fxs/fxo endpoints will be subscribed.
- Username= Gateway: account name used for subscription to the MWI server.
- EnableMWI = 1
- EnableMWISubscription = 1 (optional)
- MWIExpirationTime = 7200 (seconds)
- SubscribeRetryTime = 120 (seconds)
- MWIServerIP = xxx.xxx.xxx.xxx: IP address of the MWI server.
- MWISERVERTRANSPORTTYPE: -1 for not configured (default); 0 for udp; 1 for tcp; 2 for tls. If not configured (-1), use sip transport according to SipTransportType parameter.
- MWIOffCode: determines a digit code used by the gateway to notify the PBX that there aren't any messages for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
- MWIONCode: determines a digit code used by the gateway to notify the PBX of messages waiting for a specific extension. This code is added as prefix to the dialed number. The valid range is a 25-character string.
- MWISuffixCode: this code, if configured is added to the generated DTMF string after dialing the extension number. For example, with Panasonic PBX, the following MWI patterns are used:
 - “MWI on” is 701<ext number>0, such as “7011010” for extension “101”
 - “MWI off” is 702<ext number>0, such as “7021010” for extension “101”
 In this example, the following parameters should be configured:
 - MWIONCode = '701'
 - MWIOffCode = '702'
 - MWISuffixCode = '0'
- ChannelSelectMode = 1: must not be set to 0 (i.e., By Phone Number).
- WaitForDialTime
- FlashHookPeriod (FXO gateways only)
- TrunkGroup and TrunkGroupSettings

3.3.2 Examples

Subscribe Example:

```
SUBSCRIBE sip:Gateway@10.33.2.10 SIP/2.0
Via: SIP/2.0/UDP 10.33.4.103;branch=z9hG4bKacAqDJNSC
Max-Forwards: 70
From: <sip: Gateway @10.33.4.103>;tag=1c133925930
To: <sip: Gateway @10.33.4.103>
Call-ID: 2658731555ISYm@10.33.4.103
CSeq: 1 SUBSCRIBE
Contact: <sip: Gateway@10.33.4.103>;expires=7200
Event: message-summary
Supported: em,timer,replaces,path
Allow: REGISTER, OPTIONS, INVITE, ACK, CANCEL, BYE, NOTIFY,
PRACK, REFER, INFO, SUBSCRIBE, UPDATE
Expires: 7200
User-Agent: Audiocodes-Sip-Gateway-MP-104 FXO/v.4.40.0.200
Content-Length: 0
```

Notify Example:

```
NOTIFY sip:Gateway@10.33.2.10 SIP/2.0
Via: SIP/2.0/UDP 10.33.2.10;branch=z9hG4bKadfe4ko
To: <sip: Gateway@10.33.4.103>;tag=1c133925930
From: <sip: Gateway@10.33.4.103>;tag=1111
Call-ID: 2658731555ISYm@10.33.4.103
CSeq: 1 NOTIFY
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY
Event: message-summary
Content-Type: application/simple-message-summary
Content-Length: 100

Messages-Waiting: yes
Message-Account: sip:401@10.33.4.103
```

MWI code examples:

- **Avaya Partner:** On:#09 Off:#10; MWIOnCode=#09 and MWIOffCode = #10
- **Avaya Merlin, Definity:** On:#53 Off:#*53
- **Mitel:** On:#09 Off:#10
- **NEC:** On:*9 Off:#9
- **Panasonic KX-TD:** On:701 Off:700
- **Toshiba DK:** On:#63 Off:#64

3.4 PBX Disconnect Supervision

This section describes PBX disconnect supervision.

3.4.1 General Disconnect Methods

The gateway supports the following methods for detecting PBX line disconnection:

- Detection of DTMF disconnect code (supported by certain PBXs).
- CAS signaling (such as receive of AB=1x from PBX, for loop start CAS signalling).
- Detection of Busy or Reorder tones (both applicable for CAS and FXO gateways).
- Detection of Current Disconnect for FXO gateways (EnableCurrentDisconnect = 1).
- Detection of Polarity Reversal for FXO (EnableReversalPolarity = 1).
- Silence detection for preconfigured timeout. The CAS gateway supports silence detection only if EnableDSPIPMDetectors = 1.
- Max Call duration.

It's possible to simultaneously configure several disconnect supervision methods.

3.4.2 DTMF Disconnect Code

This is a DTMF code sent by the PBX to the gateway notifying that the VM call is terminated. When the gateway identifies that a call is terminated by the PBX, it closes the IP session and releases the line towards the PBX. A digit pattern received from the Tel side notifies the gateway to disconnect the call.

Relevant parameter: TelDisconnectCode.

For example:

- **Panasonic:** #9; TelDisconnectCode = #9 (for Panasonic PBX)
- **Toshiba:** D; TelDisconnectCode = D (for Toshiba PBX)

3.4.3 Relevant Parameters

The relevant parameters are described as follows:

- DisconnectOnBusyTone = 1
- EnableCurrentDisconnect = 1
- CurrentDisconnectDuration = 500
- EnableReversalPolarity = 1
- EnableSilenceDisconnect = 1
- FarEndDisconnectSilencePeriod = 60 (for 60 seconds of silence)
- MaxCallDuration = 120 (seconds)
- TelDisconnectCode = '#9'
- EnableDSPIPMDetectors = 1 (**Note:** Applicable for CAS gateways. Must be enabled in the gateway Feature Key. The number of gateway channels may be reduced -- please refer to the User's Manual.)

3.5 Calls from Voice Mail to PBX via Analog FXO Gateway

When the VM initiates a call to the PBX via the gateway, the gateway receives an INVITE message and uses the User Part of the request URI as a called number. The FXO gateway seizes one of its available lines, waits for dial tone detection, and then dials the DTMF digits.

There is also an option to configure the gateway to dial without waiting for a dial tone (`IsWaitForDialTone = 0`), instead it waits for `WaitForDialTime` before dialing the digits. The gateway sends a 180 response immediately after receiving the INVITE message.



Note: It's also possible to configure FXO to use early media, and send 183 + SDP instead of 180. To implement this, set `EnableEarlyMedia` to 1 and `ProgressIndicator2IP` to 1.

The subsequent gateway behavior depends on whether or not the Answer Supervision feature is enabled. If Answer Supervision is enabled, the gateway detects when the remote party answers the call.

Currently, there are four answer supervision methods:

- Detection of Polarity Reversal signal sent by the PBX to the gateway over the analog line
- Speech Detection of the answering party (if the party says “hello” or any other word)
- Detection of fax or modem answer tone
- DTMF code sent by the PBX

If Answer Supervision is not used (`EnableReversalPolarity = 0`, `EnableVoiceDetection = 0`, and no DTMF Connect code), the FXO gateway sends a 200 OK response immediately after completing the dialing. If Answer Supervision is used, a 200 OK + SDP are later sent only if the remote party answers the call.

3.5.1 Relevant Parameters

The relevant parameters are described as follows:

- `IsTwoStageDial = 0`
- `EnableVoiceDetection = 1`
- `TELConnectCode = 'xxx'`: defines the DTMF sequence sent by the PBX if the remote party answers the call. This is supported by some PBXs.
- `ChannelSelectMode = 1`: activates the 'Cyclic ascending' channel selection mode.
- Additional related parameters:
 - `EnableEarlyMedia`
 - `ProgressIndicator2IP`
 - `EnableReversalPolarity`
 - `IsWaitForDialTone`
 - `WaitForDialTime`
 - DTMF dialing parameters: `DTMFDigitLength`, `DTMFInterDigitInterval`, `DTMFVolume`
 - `TrunkGroup` and `TrunkGroupSettings`: assign local numbers to gateway endpoints and define port allocation method per trunk group.

3.6 Calls from Voice Mail to PBX via Digital CAS Gateway

The T1 Loop Start access lines are used for connection between the PBX and Voice Mail (or gateway). The PBX typically uses the Foreign Exchange Office (FXO) interface type while the VM/gateway uses the Foreign Exchange Station (FXS) interface type. The disadvantage of loop-start signaling is the inability to be notified when a remote party answers the VM-to-PBX call (answer supervision). Some PBXs notify the VM that the call is answered by sending a special DTMF code.

Currently, there are three answer supervision methods:

- Speech Detection of the answering party (if the party says “hello” or any other word)
- Detection of fax or modem answer tone
- DTMF code sent by the PBX

When the VM initiates a call to the PBX via the gateway, the gateway receives an INVITE message and uses the User Part of the request URI as a called number. The gateway seizes one of its available T1 lines, waits for dial tone detection, and then dials the DTMF digits. There is also an option to configure the gateway to dial without waiting for the dial tone (by using a different CAS table). If EarlyMedia feature is enabled, the CAS gateway sends a 183 + SDP response immediately after completing the dialing. If EarlyMedia is disabled, the gateway sends instead a 180 Ringing response.

The subsequent gateway behavior depends on the used CAS table and whether or not the Speech Detection feature is enabled. If EnableVoiceDetection is set to 1, and EnableDSPIPMDetectors is set to 1, the gateway detects the answering party (if the party says “hello” or any other word), and then sends 200 OK + SDP. If voice detection is disabled, 200 OK + SDP is sent only after some timeout as defined in the CAS table (default is 20 sec).



Notes:

- **To enable voice detection**, the gateway Feature Key should contain the “IPMDetector” DSP option.
- **Setting** EnableDSPIPMDetectors to 1 can reduce the number of gateway channels (refer to the User’s Manual).

3.6.1 Relevant Parameters

The relevant parameters are described as follows:

- EnableVoiceDetection = 1
- EnableDSPIPMDetectors = 1
- WaitForDialTime = 2000
- TELConnectCode = ‘xxx’: defines the DTMF sequence sent by the PBX if the remote party answers the call. This is supported by some PBXs.
- Additional related parameters:
 - EnableEarlyMedia
 - DTMF dialing parameters: DTMFDigitLength, DTMFInterDigitInterval, DTMFVolume
 - TrunkGroup and TrunkGroupSettings: assign local numbers to gateway endpoints and define port allocation method per trunk group

4 QSIG, Euro ISDN, and NI2 Operation

This section describes the QSIG, Euro ISDN, and NI2 Operations.

4.1 Calls from PBX to Voice Mail

The QSIG protocol is based on ECMA-165, and ECMA-174 (Call Diversion Supplementary services) standards. The Euro ISDN protocol is based on ETS 300 102-1 and ETSI EN 300 207-1 (Diversion Supplementary Services). The NI2 protocol is based on SR4994 and GR-2942 "Primary Rate Interface Generic Requirements for Message Waiting Notification". When the PBX makes a call to the VM, it sends a Q.931 SETUP message. The SETUP message contains various Information Elements (IE) such as Calling Party number, Called Party Number, and Facility IE.

The call can be forwarded to VM due to the following reasons: Call Forwarding Unconditional, Call Forwarding Busy, Call Forwarding No Reply, and Call Deflection. The Facility IE of the forwarded call contains a DivertingLegInformation2 with Diverting Number and Diversion Reason (according to IETF draft draft-levy-sip-diversion-08). The gateway interworks these two parameters to SIP Diversion header as a Tel URI or as a SIP URI (configurable using UseSIPURIForDiversionHeader) in the INVITE message that is sent to the VM. It is also possible to configure the gateway to include "target" and "cause" parameters in the INVITE message according to RFC 4458 (Voicemail URI). This option is enabled by setting the *ini* file parameter EnableVMURI to 1.

When using QSIG, Euro ISDN, or NI2 protocols to interface between the PBX and VM, there is no need to use SMDI, as these protocols can provide all necessary information about the VM call. However, if necessary, the SMDI can be used with ISDN protocols.

4.1.1 Relevant Parameters

The relevant parameters are described as follows:

- **UseSIPURIForDiversionHeader:** defines the URI format in the Diversion header.
 - 0 = 'tel:' (default), e.g., 'tel:0000066242'
 - 1 = 'sip:' (e.g., 'sip:0000066242@10.2.3.4')
- **EnableVMURI:** if enabled, the INVITE includes "target" and "cause" parameters according to RFC 4458 (Voicemail URI)
- Tel to IP routing and manipulation parameters.
- **TrunkGroup and TrunkGroupSettings:** assign local numbers to gateway endpoints and define port allocation method per trunk group.
- **ECNLPMODE = 1:** Disables Echo Canceller NLP. This can improve the gateway's detection of DTMF digits during the playing of IVR announcements.
- **EchoCancellerAggressiveNLP = 0.**

4.2 Transfer Request from the IP Voice Mail to PBX

After a call from the PBX to the VM is established (as described in Section 3.1), the VM may resolve to transfer the call to an operator or to another PBX extension (or even to an external number).

The Transfer request is sent by the IP VM using a REFER message. If the IP address in the Refer-To URI is the address of the gateway itself, the gateway initiates a call transfer to the PBX extension number. If the IP address in the Refer-To URI is different from the address of the gateway, the gateway establishes a new SIP connection to that address. (**Note:** If FQDN is used in the Refer-To header instead of in the gateway numerical IP address, you should configure in the gateway's Internal DNS table the association between this FQDN and 127.0.0.1.)

- **TrunkTransferMode_x = 0:** the QSIG/EuroISDN gateways perform a 'Transfer' by making a Tandem (Hair pinning) call. After the gateway receives a SIP REFER message, it initiates a new Call-to-PBX party according to Refer-To URI. The original SIP call between the gateway and the VM is released after the PBX answers the call.
- **TrunkTransferMode_x = 2:** the gateway performs EuroISDN ECT transfer/QSIG Path Replacement transfer (ECMA-176) if supported by the switch. With QSIG protocol, the gateway first performs a Tandem transfer, and updates the switch on completion of this transfer. A few seconds later, the switch should propose to start the Path Replacement. The gateway performs it and disconnects the TDM call legs. The QSIG switch should be configured to support this feature.
- **TrunkTransferMode_x = 3 and LineTransferMode = 1:** When a SIP REFER message is received, the ISDN gateway waits for WaitForDialTime, dials the Refer-To number, and then releases the call.
- **TrunkTransferMode_x = 3 and LineTransferMode = 2:** The ISDN gateway waits for WaitForDialTime and dials the Refer-To number. If during WaitForBusyTime no busy or reorder tones are detected, the gateway completes the call transfer by disconnecting the call. If during WaitForBusyTime the gateway detects busy or reorder tones, the transfer operation is cancelled and the gateway sends a SIP NOTIFY message with a failure reason in the NOTIFY body (such as 486 if busy tone was detected).
- **TrunkTransferMode_x = 4:** the gateway performs QSIG Single Step call transfer (ECMA-299). Currently, most switches don't support this procedure.
- **TrunkTransferMode_x = 5:** IP-to-Tel Blind Transfer "inband" mode supported for ISDN (PRI/BRI) protocols. Implemented according to AT&T Toll Free Transfer Connect Service (TR 50075) "Courtesy Transfer-Human-No Data", but is supported for all ISDN variants. When the gateway receives a SIP REFER message, it performs a blind transfer by first dialing the DTMF digits (transfer prefix) defined by the parameter XferPrefixIP2Tel (configured to "*8" for AT&T service), and then (after 500 msec) the gateway dials the DTMF of the number (referred) from the Refer-To header sip:URI userpart.

If the host part of the Refer-To sip:URI contains the gateway's IP address, and if the Trunk Group selected according to the IP to Tel Routing table is the same Trunk Group as the original call, then the gateway performs the in-band DTMF transfer; otherwise, the gateway sends the INVITE according to regular transfer rules.

After completing the in-band transfer, the gateway waits for the ISDN Disconnect message. If the Disconnect message is received during the first five seconds, the gateway sends a SIP NOTIFY with 200 OK message; otherwise, the gateway sends a NOTIFY with 4xx message.

- **TrunkTransferMode_x = 6:** Supports AT&T Toll Free Out-Of-Band Blind Transfer for Trunks configured with the 4ESS ISDN protocol. AT&T Courtesy Transfer is a supplementary service which enables a user (e.g., user "A") to transfer an established call between it and user "B" to a new call between users "B" and "C", whereby user "A" does not have a call established with user "C" prior to call transfer. The gateway handles this feature as follows:
 - **IP-to-Tel (user side):** When a SIP REFER message is received, the gateway initiates a transfer by sending a Facility message to the PBX.
 - **Tel-to-IP (network side):** When a Facility message initiating an out-of-band blind transfer is received from the PBX, the gateway sends a SIP REFER message to the IP side (if the EnableNetworkISDNTransfer parameter is set to 1).

4.2.1 Relevant Parameters

The relevant parameters are described as follows:

- EnableTransfer = 1: must be set to 1 to enable transfer.
- TrunkTransferMode_x can equal the following:
 - **0:** Use tandem (hair pinning) call transfer
 - **2:** QSIG Path Replacement transfer or Euro ISDN ECT transfer (EN 300 369-1) or NI2 TBCT
 - **3:** Blind and SemiBlind Xfer
 - **4:** QSIG Single Step transfer
 - **5:** InBand Isdn Xfer
 - **6:** 4ESS OutOfBand Blind Xfer
- LineTransferMode can be set to the following:
 - **0:** None
 - **1:** Blind = PBX blind transfer
 - **2:** Semi Supervised = PBX semi-supervised transfer
- QsigCallTransferReverseEndDesignation is relevant only for QSIG:

Currently, during blind transfer, when the QSIG gateway receives sip:REFER, it sends an INVITE to itself according to Refer-To header. The gateway then places a second call to the PBX. When the call is answered, the gateway sends a Facility message with CallTransferComplete (Primary) APDU to the PBX on the first QSIG call leg. Nortel CS1K PBX waits for CallTransferComplete (Secondary) APDU to send Facility with CallPropose APDU, as part of QSIG Path Replacement procedure. To resolve this problem, set the above parameter to 1 which enables the gateway to also send a Facility message with CallTransferComplete (Secondary) APDU on the second call leg, similar to consultation transfer.

Thus, if the parameter is set to 1:

 - **For blind transfer:** The primaryLeg sends CallTransferComplete with EndDestination set to SecondaryEnd
 - **For consulting transfer:** The primaryLeg sends CallTransferComplete with EndDestination set to SecondaryEnd, and SecondaryLeg sends CallTransferComplete with EndDestination set to PrimaryEnd

4.3 MWI Notification from IP Voice Mail to PBX

The IP VM uses SIP NOTIFY messages to inform the gateway of Message Waiting Indications (MWI). The gateway may subscribe to a VM server (a single subscription per gateway) using the SIP MWI package (RFC3842 Message Summary and Message Waiting Indication Event Package for SIP).

After a successful subscription, the VM server can start sending NOTIFY messages to the gateway with MWI. The extension number sent to the PBX is provided in the Message Account header in a NOTIFY message, as shown in the example below with extension number 401:

```
Messages-Waiting: yes
Message-Account: sip:401@10.13.77.10
Voice-Message: 2/8
```

After being informed by the VM of MWI, the QSIG gateway sends a SETUP with an Activate MWI Facility IE (Turn On) message over QSIG protocol to the PBX. After receiving the CONNECT message with MWI confirmation result from the PBX, the gateway releases the MWI call. The QSIG MWI is implemented according to the ECMA-242 standard. In this message, the gateway includes the subscriber number received in the NOTIFY message ('401' in the example above). A Deactivate MWI (Turn Off) message is used to remove the MWI. The Called Number in the MWI Q.931 SETUP can be preconfigured (DefaultNumber) or set to the same value as the subscriber number (DefaultNumber = 'ServedUser'). The Calling Number in the MWI Q.931 SETUP is empty, or it can be defined using the *ini* parameter MWISourceNumber.

For the EuroISDN protocol, the gateway sends a Facility Q.931 message containing Facility IE with a "MWIActivate" operation to the PBX or BRI phone, according to EN 300 745-1 specifications. To remove the MWI, the gateway sends another FACILITY message containing a "MWIDeactivate" operation in the Facility IE.

For the NI2 protocol, the gateway sends a Facility Q.931 message containing the Facility IE with a "messageWaitingNotificationNT2" operation to the PBX or BRI phone, according to the GR-2942 specification. To remove the MWI, the gateway sends another Facility message containing a "messageWaitingNotificationNT2" operation in the Facility IE. (The messageWaitingNotificationNT2 operation contains several arguments such as: controlType, destinationDN, mSRID, and callingNumber, which are described in detail in the GR-2942 specification). Interrogation does not exist in NI2.

For the IP-to-IP application, the device sends unsolicited SIP NOTIFY messages received from the Voice Mail server, as is to an IP Group specified by the NotificationIPGroupID parameter.

**Notes:**

- The IP to Tel Routing table can be used to route the MWI Q.931 messages to a specific Trunk Group, using regular routing rules such as according to URI user part in the From header of the received NOTIFY message.
- The gateway also accepts MWI Notify messages without Subscription to the MWI server.
- If the Message-Account parameter is missing, the gateway uses the SIP URI User Part from the To header, as the Account number.
- There is a difference between the previous QSIG MWI ECMA version and the latest one regarding coding of Invoke PDU. There are two coding options: Integer or Object Identifier.
The QSIG stack has the option to encode the operations in both ways, according to behavior bit. To use Integer coding, you need to configure ISDNIBehavior = 1073741824. Most of the PBXs support the Integer coding.
- The number of new/old voice messages received in the “Voice-Message: 2/8” parameter in NOTIFY are currently not interworked to QSIG MWI message. Number of messages in QSIG MWI is set to 1.

4.3.1 Relevant Parameters

The relevant parameters are described as follows:

- ISDNBehavior = 1073741824 (for QSIG Integer coding) or 0 (for QSIG Object Identifier coding)
- PSTNExtendedParams = 0 (default) for QSIG “Networking Extensions” format; 2 for ROSE format (according to old QSIG specifications)
- VoiceMailInterface = 3: Enables interworking of SIP NOTIFY MWI messages to QSIG Facility messages according to ECMA-242
- VoiceMailInterface = 4: A proprietary method. The gateway interworks SIP NOTIFY message to Q.931 SETUP with called number comprised of “MWIOnCode” or “MWIOffCode” plus Subscriber number (from Message-Account header), according to ‘Messages-Waiting: yes or no value’. The gateway expects to receive Q.931 RELEASE message immediately after the sent SETUP.
- VoiceMailInterface= 5: Enables interworking of SIP NOTIFY MWI messages to proprietary QSIG Facility messages for AASTRA/MATRA PBX.
- VoiceMailInterface= 6: Enables interworking of SIP Notify MWI messages to proprietary QSIG Facility messages for Siemens Hipath 4000 PBX. These MWI Facility messages contain Siemens Manufacturer Specific Information (MSI).
- VoiceMailInterface = 7: The device's IP-to-IP application is used for interworking between an IP Voice Mail server and a SIP entity. This is implemented for sending unsolicited SIP NOTIFY messages received from the Voice Mail server to an IP Group specified by the NotificationIPGroupID parameter.
- VoiceMailInterface = 8: Enables interworking of SIP NOTIFY MWI messages to EuroISDN MWI Facility (currently, only the IP-to-Tel direction is supported).
- VoiceMailInterface = 9: Enables interworking of SIP NOTIFY MWI messages to NI2 MWI Facility.
- MWIQsigMsgCentreIdPartyNumber- Message Centred Id party number
Used for QSIG MWI messages. If not configured, the parameter is not included in the MWI QSIG messages. Some PBXs (such as Siemens Hipath 4000) require configuration of this parameter. Usually, it is set to some valid number of the PBX dial plan.
- SubscriptionMode = 1 (subscription per gateway): determines the method the gateway uses to subscribe to an MWI server.
- Username= Gateway: account name used for subscription to the MWI server.
- EnableMWI = 1
- DefaultNumber: PBX identification number is included in QSIG MWI Q.931 SETUP as a called party number. If DefaultNumber = 'serveduser', then the called number in the MWI SETUP is identical to the “Served User” number (the Message-Account number as received in SIP NOTIFY message).
- MWISourceNumber: this number, if configured is used in the Q.931 MWI SETUP as a calling party number. Otherwise, the source number will be empty.
- EnableMWISubscription = 1 (optional)
- MWIExpirationTime = 7200 (seconds)
- SubscribeRetryTime = 120 (seconds)
- MWIServerIP = xxx.xxx.xxx.xxx: IP address of the VM server.
- MWIServerTransportType: -1 for not configured (default); 0 for UDP; 1 for TCP; 2 for TLS. If not configured (-1), use the SIP transport according to the SipTransportType parameter.

4.4 QSIG Interrogation

The gateway supports the interworking of QSIG Message Waiting Indication (MWI) to IP. This provides interworking between a PBX (with integrated VM) or a standalone Voice Mail system and a softswitch/application server which requires information on the number of voice mail messages waiting for a specific user. The feature also allows to interrogate PBX about MWI messages for a specific user.

4.4.1 MWI Interrogation Procedure

The MWI Interrogation procedure is as follows:

1. To invoke a QSIG interrogation, a SIP SUBSCRIBE request is sent to the gateway.
2. The SUBSCRIBE request is sent in a polling mode - the Expires header **must** be equal to zero.
3. The SUBSCRIBE To header contains the phone number to interrogate.
4. The destination Trunk Group is selected by searching the IP to Tel Routing table, using the phone number from the To header of the SUBSCRIBE message.
5. Each Trunk Group can be configured to support MWI QSIG Interrogation, using the TrunkGroupSettings_MWIInterrogationType parameter.
6. The gateway sends a QSIG SETUP message with Facility IE containing an MWI interrogation request.
7. Each such Subscribe is replied with a SIP 200 response, regardless of the PSTN response.
8. The 200 response is followed with an immediate SIP NOTIFY message with an empty body and subscribe-state terminated in order to terminate this subscribe dialog.
9. The rate of the SUBSCRIBE messages must not exceed 10 subscriptions per second.
10. Each such SUBSCRIBE invokes a QSIG interrogate request (Q.931 SETUP message with Facility IE containing an MWI Interrogation request).
11. The PBX responds by sending to the gateway a Q..931 CONNECT message containing Facility IE with an MWI Interrogation result, which includes the number of voice messages waiting for a specific user.
12. The gateway sends a SIP unsolicited NOTIFY message containing this MWI information in the message account body to an IP destination defined by the NotificationIPGroup parameter.

In addition, when a change in the status occurs (e.g., a new voice message is waiting or the user has retrieved a message from the voice mail), the PBX initiates an ISDN Setup message with Facility IE containing an MWI Activate (or Deactivate) request, which includes the new number of voice messages waiting for the user. The gateway forwards this information to the softswitch by sending a SIP NOTIFY.

Depending on the PBX support, the MWIInterrogationType parameter can be configured to handle these MWI Interrogation messages in different ways. For example, some PBXs support only the MWI Activate request (and not MWI Interrogation request). Some support both these requests. Therefore, the gateway can be configured to disable this feature or enable it with one of the following support:

- Respond to MWI Activate requests from the PBX by sending SIP NOTIFY MWI messages (i.e., does not send MWI Interrogation messages).
- Send MWI Interrogation message, but don't use its result. Instead, wait for MWI Activate requests from the PBX.
- Send MWI Interrogation message, use its result, and use the MWI Activate requests.

4.4.2 Parameters for Configuring the QSIG MWI Interrogation Feature

The following parameters are used to configure the QSIG MWI Interrogation feature:

- ProtocolType = 21 or 23 (E1 or T1 QSIG)
- ISDNBehavior = 1073741824 (for Integer coding) or '0' for Object Identifier coding)
- EnableMWI = 1
- VoiceMailInterface = 3 (QSIG)
- NotificationIPGroupID = 1 (IP group ID where the gateway sends SIP MWI NOTIFY messages).



Note: To define an IP Group for notifications, you need to configure the IPGroup and ProxySet parameters.

- Trunk Group Settings: set the MWI Interrogation Type (0-3):
 - 0: None (disabled)
 - 1: Use Activate Only - don't send any interrogation messages, but respond to PBX MWI Activate messages and send Notify messages
 - 2: Interrogate no result - send MWI interrogation messages, but don't send any Notify for the interrogation result response.
 - 3: Interrogate use result - full feature, use interrogate result for Notify messages as well (in addition to MWI Activate).



Note: This parameter can also be configured in the Trunk Group Settings table of the Web interface.

For example:

```
[ TrunkGroupSettings ]
FORMAT TrunkGroupSettings_Index =
TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode,
TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup,
TrunkGroupSettings_MWIInterrogationType;
TrunkGroupSettings 0 = 3, 1, 255, , , -1, 3;
[ \TrunkGroupSettings ]
```

A SMDI Examples (Bellcore)

SMDI Message (No Answer):

```
MD0010003N0000066242 0000061382
```

SMDI Message Interpretation (No Answer):

```
MessageDeskNum 001 PositionNum 0003 CallType N ForwardDN
0000066242 CallingDN 0000061382 CallingDNStatus CallingName
```

Corresponding INVITE Message (No Answer):

```
INVITE sip:0010003@157.55.89.37;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacZjvZlHl;alias
Max-Forwards: 70
From: <sip:0000061382@10.197.96.30>;tag=1c2794514316
To: <sip:0010003@157.55.89.37;user=phone>
Call-ID: 2415530342UgsW@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:0000061382@10.197.96.30;transport=tcp>
Supported: em,100rel,timer,replaces,path
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,
SUBSCRIBE,UPDATE
Diversion: <tel:0000066242>;reason="no-
answer";screen=no;privacy=off
Content-Type: application/sdp
Content-Length: 246

v=0
o=AudiocodesGW 296442 595422 IN IP4 10.197.96.30
s=Phone-Call
c=IN IP4 10.197.96.30
t=0 0
m=audio 6020 RTP/AVP 0 8 96
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
a=sendrecv
```

A.1 Always Forward Call

SMDI Message (Always Forward Call):

```
MD0010003A0000066259 0000061382
```

SMDI Message Interpretation (Always Forward Call):

```
MessageDeskNum 001 PositionNum 0003 CallType A ForwardDN
0000066259 CallingDN 0000061382 CallingDNStatus CallingName
```

Corresponding INVITE Message (Always Forward Call):

```
INVITE sip:0010003@157.55.89.37;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKacWPlxbnR;alias
Max-Forwards: 70
From: <sip:0000061382@10.197.96.30>;tag=1c843327572
To: <sip:0010003@157.55.89.37;user=phone>
Call-ID: 646011738XjNZ@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:0000061382@10.197.96.30;transport=tcp>
Supported: em,100rel,timer,replaces,path
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,
SUBSCRIBE,UPDATE
Diversion:<tel:0000066259>;reason="unconditional";screen=no;priv
acy=off
Content-Type: application/sdp
Content-Length: 246

v=0
o=AudiocodesGW 688644 396644 IN IP4 10.197.96.30
s=Phone-Call
c=IN IP4 10.197.96.30
t=0 0
m=audio 6020 RTP/AVP 0 8 96
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
a=sendrecv
```

A.2 Direct Call

SMDI Message (Direct Call):

```
MD0010003D
```

SMDI Message Interpretation (Direct Call):

```
MessageDeskNum 001 PositionNum 0003 CallType D ForwardDN
CallingDN CallingDNStatus CallingName
```

Corresponding INVITE Message (Direct Call):

```
INVITE sip:0010003@157.55.89.37;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.197.96.30;branch=z9hG4bKaceXAMiMY;alias
Max-Forwards: 70
From: <sip:1@10.197.96.30>;tag=1c214321282
To: <sip:0010003@157.55.89.37;user=phone>
Call-ID: 1519122287AUgC@10.197.96.30
CSeq: 1 INVITE
Contact: <sip:1@10.197.96.30;transport=tcp>
Supported: em,100rel,timer,replaces,path
Allow:
REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,
INFO,SUBSCRIBE,UPDATE
Content-Type: application/sdp
Content-Length: 246

v=0
o=AudiocodesGW 197755 607553 IN IP4 10.197.96.30
s=Phone-Call
c=IN IP4 10.197.96.30
t=0 0
m=audio 6020 RTP/AVP 0 8 96
a=rtpmap:0 pcmu/8000
a=rtpmap:8 pcma/8000
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=ptime:20
a=sendrecv
```

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