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GVP Deployment Guide

How the PSTN Connector Works

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Read here about how the Public Switched Telephone Network (PSTN) Connector performs its role in a GVP deployment.

Operational Overview

The GVP PSTN Connector is a network layer element which provides access to the core presentation layer services by using SIP. External TDM networks access the PSTN Connector through Dialogic Application Programming Interfaces (API) by using E1 Channel Associated Signaling (CAS), T1 CAS, and T1/E1 ISDN signaling protocols.

The PSTN Connector leverages subscribed transfer services, and provides many advanced inbound and outbound-calling features to support integration with TDM networks.

For a description of how the PSTN Connector processes inbound and outbound call triggers, see Basic PSTN Call Flow (Inbound) and Basic PSTN Call Flows (Outbound).

Signaling Protocols

The PSTN Connector provides interfaces for three signaling protocols Integrated Services Digital Network (ISDN), Robbed-bit signaling (RBS), and Channel Associated Signaling (CAS).

ISDN PRI

- ISDN Primary Rate Interface (PRI) provides different service offerings depending on the geographic region, for example:
  - In North America and Japan 23 B channels and 1 D channel, yield a total bit rate of 1.544 Mbps (the PRI D channel runs at 64 kbps).
  - In Europe, Australia, and other parts of the world 30 B channels and 2 64-kbps D channels yield a total bit rate of 2.048 Mbps.

The PSTN connector supports these ISDN PRI T1 and E1 protocol variants:

T1-ISDN PRI	E1-ISDN PRI
• 4ESS ISDN (tested with AT&T 4ESS switch) • NI-2 ISDN • NT-1 ISDN • 5ESS ISDN (tested with Lucent 5ESS switch) • Nortel Custom ISDN (test with Nortel DMS-100 switch)	• NE1 ISDN • NET5 ISDN • CTR4 ISDN • QSIG ISDN

For a complete list of supported specifications and standards, including ISDN PRI physical layer, see Specifications and Standards.

Robbed-Bit Signaling

- Robbed-bit Signaling (RBS) is a type of CAS that is sometimes referred to as in-band signaling. CAS signals each traffic channel instead of a single dedicated channel (like ISDN). The signaling associated with a traffic circuit is permanently associated with it. The most common types of CAS are loop start, ground start, Equal Access North American (EANA), and E&M (Ear & Mouth).

CAS also processes the receipt of DNIS and ANI information, which is used to support authentication and other functions. The PSTN Connector can be configured to restrict or pre-define the length of the DNIS and ANI on an incoming call.

The PSTN Connector supports these types of RBS:

• Line-side T1 • Group A T1 • Group B T1 • Group D T1	• wink start T1 • ground start T1 • loop start T1 • immediate start T1

Channel Associated Signaling

- The PSTN Connector supports standard E1 CAS. CAS transmits signaling information within the voice channel. CAS is configured on an E1 controller and enables the access server to send or receive analog calls. It is categorized as an out-of-band signaling method because it uses the 16th channel (or time slot).

Transfer Services and Features

The PSTN Connector supports transfer services and features in various ways. This section describes how the PSTN functions with each of these transfer types.

Dialogic Transfers

- Blind transfer The PSTN Connector performs a Dialogic Blind or Hook Flash transfer when it receives a REFER message from the Media Control Platform (through the Resource Manager) and releases all ports associated with the call. If the call is not answered or busy, it is usually routed back to GVP. By releasing the call to the PBX, the PSTN Connector is free to handle new calls without handling the transferred call progression.

- Bridge transfer The PSTN Connector supports Dialogic time slot bridging. In this case, when the two separate call legs are established between the PSTN Connector and the Media Control Platform, the media flow bridging occurs at the PSTN Connector. This prevents the latency that is generated when the media is passed to and from the Media Control Platform and allows the Full Call Recording (FCR) of media when it is passed to the Media Control Platform.

For more information about FCR, see Media Services.

AT&T Transfer Connect

- GVP acts as the redirecting party (RP), which sends calls to other locations or target parties (TP). AT&T service supports in-band (IB) and out-of-band (OOB) invocation or triggers. For OOB triggers and data forwarding, the PSTN Connector and TP must have ISDN PRI, as OOB signaling occurs on the D-channel. Transfers or redirection can be provisioned for OOB triggers only, or for OOB triggers and OOB data forwarding.
    - In-band GVP requests the redirection of an answered call by out-pulsing an in-band DTMF touch-tone command (or trigger). The call is then placed on hold by the network. The PSTN Connector role is limited to passing the in-band trigger and dial string received from Media Control Platform to the network. The in-band transfer processing logic lies with the Media Control Platform component.
    - Out-of-band When an inbound or outbound call is established between the PSTN Connector and the AT&T network, the PSTN Connector initiates a redirection FACILITY message on the signaling channels which then disconnects the call from GVP and transfers it to the TP (regardless of the outcome of the call). The FACILITY message contains User-to-User Information (UUI) from GVP, which is passed to the TP through the network. The network send notification back to GVP in another FACILITY message.

For more information about UUI, see Inbound-/Outbound-Calling Features on page 116. To find out how Media Control Platform works with AT&T Transfer Connect, see Transfer Methods for AT&T Transfer Connect.

Two B-Channel Transfer

When a controller (or subscriber) uses more than one PRI, and transfers two calls that are using different PRIs (each controlled by its own D-channel), the controller must obtain a PRI identifier for the PRI of one of the two calls before it can request the transfer. TBCT can also send transfer notifications to the two callers, but this is optional.

The TBCT implementation is defined in Telcordia Technologies Generic Requirements GR-2865-CORE. For a complete list of supported specifications and standards, see Specifications and Standards.

Release Link Trunk Transfer

- The initial calls can be on the same or different PRI trunk groups, but if they are on different ones, they must connect to the same DMS switch.

Both the primary and secondary trunks must be configured with the RLT feature enabled. If an outbound call from the GVP is redirected to a third-party number, then that number must also be configured for RLT. If the third-party number is not configured for RLT, the switch cannot return a call ID to GVP, and the calls cannot be transferred.

Explicit Call Transfer

- ECT uses two lines to transfer a call, a primary line and an outbound line. When the outbound call reaches the alerting (ringing) state, GVP sends a request to transfer the call to the switch. When the switch accepts the request, the user is released from the calls and they are connected directly. The transfer can be configured to wait until the outbound call is answered before initiating.

Q.SIG Call Transfer

- Q.SIG uses a method of call control in which the switch-type is defined in the configuration file, because each switch vendor uses a different method of implementing the service. This method of call control is defined in EN 300 171 and EN 300 172.

Path-Replacement is a supplemental service that uses two lines, a primary line and an outbound line. Once a switch has accepted the transfer request, both calls are connected at the switch and the GVP releases both B channels. The Path-Replacement method of call transfer is defined in the ETS 300 258 and ETS 300 259 ETSI Specification.

Selected Features

This section describes some of the advanced features and functionality that are supported by the PSTN Connector for inbound and outbound calling:

Inbound-/Outbound-Calling Features

CTI Connector Integration

- The PSTN Connector supports CTI integration with IVR Server in front or behind the switch, and with GVPi and NGI. It passes Dialogic port information to the CTI Connector in SIP custom headers.

Port Management

- PSTN Connector ports can be configured to accept inbound calls, outbound calls, or both. When the Type parameter is set to In/Out and glare occurs, priority is given to inbound calls and outbound calls are given a predefined number of retries.

User-to-User Information

- ISDN PRI User-to-User information enables a user to send information to the network which can then be transferred to a remote user. GVP can send or receive this UUI in a single call leg or transfer it end-to-end from the * incoming call to the outbound call during a transfer. The PSTN Connector transmits UUI from incoming and outbound calls by using the SIP `X-Genesys-GVP-UUI` custom header in the INVITE or REFER (if transferred) messages to and from the Media Control Platform.

The PSTN Connector supports ISDN codeset when UUI is propagated during a transfer. The code set mechanism enables different geographic areas to use their own nation-specific information elements within the data frames.

Presentation and Screening Indicators

- When the call is inbound, the PSTN Connector extracts the Presentation and Screening Indicators, Numbering Plan, and Number Type from the Calling Party Number IE of the ISDN call set-up message if it is supported by the network. The PSTN Connector then propagates this information to the Media Control Platform in the Remote-Party-ID header of the INVITE message. When the call is outbound, this information is extracted from the outbound SIP INVITE message that is sent by the Media Control Platform and the PSTN Connector updates the IE appropriately.

AT&T Coda Extensions

- When a call is inbound from an AT&T network, the PSTN Connector extracts Billing Number and Information Indicator Digits from the ISDN call set-up message and propagates this information to the Media Control Platform in the `X-Genesys-ATT-CODA` custom header of the SIP INVITE message. When the call is outbound, the PSTN Connector extracts this information from the outbound INVITE custom header that is sent by the Media Control Platform and propagates it to the TDM network.

Inbound-Calling Features

ISDN Alerting

- This calling feature is enabled (by default) during inbound call setup to avoid delays in answering calls. The PSTN Connector can be configured to enable or disable this functionality by setting the DisableISDNAlerting parameter to True or False.

Overlap Receive DNIS/ANI

- Some PSTN network switches send DNIS and ANI in overlap-in-band mode. After the call is established, the PSTN Connector waits for this information for a predefined period of time and sends it on to the Media Control Platform (through Resource Manager) in a SIP INVITE message to initiate a dialog. This functionality can be configured for T1 RBS, T1-ISDN, E1 CAS, or E1-ISDN by setting the OverlapReceivedEnable to True or False. It is disabled by default.

Redirecting Number from IE

- The PSTN Connector extracts the Redirecting Number (RN) from the RN Information Element (IE) of an inbound ISDN call set-up message if it is supported by the network. The RN IE identifies the number from which a call is diverted or transferred. This feature is optional and controlled by the network. The PSTN Connector extracts the RN, reason, and original called * number (OCN) and propagates this information to the Media Control Platform by using the History-Info header of the SIP INVITE message. If the RN IE is not available, the PSTN Connector returns an empty string.

Outbound-Calling Features

Disconnect Cause Propagation

- When the PSTN Connector disconnects a call from GVP, it propagates the cause to the PSTN network by using a SIP error response code For example, ISDN=111 (Protocol Error) is propagated with a SIP 400 (Bad Request) response code. For a list of ISDN DISCONNECT messages that map to SIP error response codes, see the Genesys Voice Platform User s Guide.

Call-Progress Analysis

- During outbound-call initiation, the PSTN Connector might receive a request from the remote party to detect call-progress events on the media stream. If this occurs, the Media Control Platform enables Call Progress Analysis (CPA) and responds with a SIP 200 OK message which contains a list of supported events (in the X-Detect header) that can be detected. The PSTN Connector then sends notification of detected events to the Media Control Platform in SIP INFO messages.

The newly defined X-Detect SIP header in the INVITE (200 OK) message indicates a request or response. The request includes a list of event types that the remote party wants notification of, for example, X-Detect:Request=CPT,FAX. The response includes a list of event types that the PSTN Connector is able to detect, for example, X-Detect:Response=CPT,FAX. If the X-Detect header is not in the request the PSTN Connector proceeds as though CPA is not required.

The X-Detect header can only be used while the SIP dialog is being established. After that, detection capabilities are determined and cannot be changed.

Table: CPA Categories and Sub-types as Supported with Dialogic

<table>
<tr><th>Type</th><th>Subtype</th><th>Previous Subtype Name (for backward compatibility)</th><th>Supported with Dialogic</th><th>Supported with CPD Library</th></tr>
<tr><td>AMD</td><td>AUTOMATA</td><td>Automatic</td><td>Yes</td><td>Yes</td></tr>
<tr><td rowspan="9">CPT</td><td>NoRingBack</td><td>No Ring Back</td><td>Yes</td><td>Yes</td></tr>
<tr><td>BUSY</td><td>Busy</td><td>Yes</td><td>Yes</td></tr>
<tr><td>SIT-RO</td><td>ReOrder</td><td>Yes</td><td>Yes</td></tr>
<tr><td>Not-In-Service</td><td>Not-in-Service</td><td>No</td><td>Yes</td></tr>
<tr><td>SIT-IC</td><td>Operator Intercept</td><td>Yes</td><td>Yes</td></tr>
<tr><td>NoDialTone</td><td>No Dial Tone</td><td>Yes</td><td>Yes</td></tr>
<tr><td>UnAllocatedNumber</td><td>UnAllocated Number</td><td>No</td><td>Yes</td></tr>
<tr><td>SIT-VC</td><td>Vacant Circuit</td><td>Yes</td><td>Yes</td></tr>
</table>

Type	Subtype	Previous Subtype Name *(for backward compatibility)*	Supported with Dialogic	Supported with CPD Library
	SIT	Unknown SIT	No	Yes
	NoAnswer	NoAnswer	Yes	Yes
FAX	CED (FAX1)	CED (FAX1)	Yes	Yes
	CNG (FAX2)	CNG (FAX2)	Yes	Yes
PVD	VOICE	Voice	Yes	Yes
	Cadence	Cadence	Yes	Yes
	LoopCurrent	LoopCurrent	Yes	Yes