GVP Deployment Guide

PSTN Connector

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PSTN Connector

The PSTN Connector is a stand-alone component that provides connectivity to traditional telephony networks and equipment, such as a private branch exchange (PBX) or automatic call distribution (ACD). For existing deployments that use Dialogic TDM cards, the PSTN Connector provides seamless integration and migration to the IP-based GVP 9.0 architecture.

The PSTN Connector supports inbound and outbound calling by acting as a border element, interfacing with a PSTN cloud or PBX or ACD on one side, and through SIP Server, interacting with the Media Control Platform through the Resource Manager on the other.

This section provides an overview of the following topics:

PSTN Connector Roles

The PSTN Connector acts as a media gateway by using Dialogic boards to interface with the TDM side of the network and translate TDM calls to SIP calls so they can be handled by SIP Server and the GVP components.

PSTN Connector Functions

PSTN Connector functions are primarily based on GVP 7.x Voice Communication Server (VCS), but with a few enhancements. The VoIP interface is compliant with RFC 3261 for SIP session control and with RFC 3550 for packetization and control of RTP packets.

The PSTN Connector performs the following functions:

- Captures and transmits DTMF tones (using Dialogic technology) to and from the TDM networks.
- Receives and controls ISDN and non-ISDN calls from TDM networks.
- Detects and emits DTMF tones to and from the Media Server over VoIP networks
- Converts TDM signals and media to SIP messages and RTP over VoIP networks.
- Works with the Media Server to provide media playback and buffer management.
- Provides Dialogic port management and the ability to re initialize ports that are stuck (by using Genesys Administrator).

- Captures dynamic call, port, and Dialogic board statistics in SNMP MIB tables, and generates traps for critical information or failures.
- Passes Dialogic port numbers on to the CTI Connector in SIP custom headers to facilitate integration with IVR Server.
- Supports User-to-User Information (UUI) messages by using the codeset that is sent from a user to the network to transfer information to another remote user.
- Provides bidirectional port functionality and a strategy for managing glare.
- Supports Call Progress Analysis for outbound calls using PSTN Connector.
- Provides media prefilling.
- Supports features for inbound-call support, such as:
    - Disable ISDN Alerting and Overlap Receive DNIS/ANI.
    - Extracting data such as, Redirecting Number (RN), Presentation and Screening Indicators, Numbering Plan, Numbering Type, Billing Number, and Information Indicator Digits from ISDN Information Elements (IE).
- Supports features for inbound and outbound call support, such as:
    - Disconnect Cause Propagation

For more information about how the PSTN Connector performs its functions, see How the PSTN Connector Works.

PSTN Connector Interfaces

The PSTN Connector provides interfaces to support the following three signaling protocols:

1. Integrated Services Digital Network (ISDN) Comprised of digital telephony and data-transport services offered by regional telephone carriers. ISDN uses digitization to transmit voice, data, text, graphics, music, video, and other source material over existing telephone wires.
2. Robbed-bit signaling (RBS) A specific type of Channel Associated Signaling (CAS), which robs the least significant bit of information in a T1 signal from the channels that carry voice and uses it to transmit framing and clocking information.
3. CAS for channelized E1 lines Commonly used in Latin America, Asia, and Europe and is configured to support channel banks in the network that convert various battery and ground operations on analog lines into signaling bits, which are forwarded over digital lines.

To find out how these signaling protocols interacts with the PSTN Connector, see How the PSTN Connector Works.

PSTN Connector Supported Transfers

The PSTN Connector supports many types of call transfers for both inbound and outbound calls, including the following types of transfers:

- **Dialogic Transfers:** Dialogic blind and bridge transfers are treated the same as any other blind and bridge transfers within GVP. See Transfer Types.

Subscribed Transfer Services

- **Transfer Connect** is an AT&T service which enables subscribers to redirect or transfer a call to another location or target party (TP). The toll-free subscriber that receives and transfers the incoming call is referred to as the redirecting party (RP)--in this case, GVP. This service supports data forwarding for inbound and outbound calls, and in-band and out-of-band transfers.
- **Two B-Channel Transfer** (TBCT) service enables a controller (or subscriber) on a PRI to request the Stored Program Control Switch (SPCS) to connect two independent calls on the controller s interface--in this case, the PSTN Connector. When the SPCS accepts the request, the controller is released and the two calls are connected directly.
- **Release Link Trunk Transfer** (RLT) call transfer accepts calls on two different B-channels, pulls the calls back from the GVP, and bridges them at the switch. It then releases both B-channels for further inbound or outbound calls. RLT works with Nortel DMS-100 and DMS-250 switches on ISDN PRI T1 trunk groups.
- **Explicit Call Transfer** (ECT) enables an ISDN PRI user (in this case, GVP) to send requests to the switch to connect two independent calls on the users interface. The two calls can use the same PRI trunk or different PRI trunks. GVP implements a supplemental ECT service, as defined in EN 300 367 and EN 300 369-1. ECT, and supports the ECT_AUS, ECT_UK, and ECT_NZ variants.
- **SIG Call Transfer** Q Signaling (Q.SIG) is a signaling protocol that uses Remote Operation Support Element (ROSE) encoding and object identifiers to provide various supplementary services, including transfers, call control, and path-replacement. The GVP implementation conforms to the method recommended by the European Telecommunications Standard Institute (ETSI) and is based on the ITU-T Q.931 standard. Although Q.SIG is not technically a subscribed service, GVP must be on a network that supports Q.SIG to access its call control and transfer features. In addition, the following conditions must be met:
    - The two connected calls must have compatible B-channels.
    - Both incoming and outgoing calls from the PSTN Connector must be answered.

To use TBCT, RLT, or ECT you must subscribe to the service and the following conditions must be met:

- The two connected calls must have compatible B-channels.
- One of the two calls must be answered.
- If the other call is outgoing (from PSTN Connector), it can be answered or alerting.
- If the other call is incoming (to PSTN Connector), it must be answered.

For more information about how the PSTN Connector integrates with the PSTN network to utilize these transfer services and signaling protocols, see How the PSTN Connector Works.

> **Tip**
>
> The PSTNC is only available on the GVP 8.1.4 CD, but it functions properly with GVP

8.1.6 and above.