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DEFINITY® Communications System

Generic 3V4

CallVisor® ASAI Technical Reference

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About This Document

This reference manual provides detailed information regarding the CallVisor Adjunct/Switch Application Interface[1] for the Generic 3 switch.

ASAI is a communications interface that allows application processors (called adjuncts in this document) to access switch features and control switch calls.

The ASAI is implemented using either an Integrated Services Digital Network (ISDN) Basic Rate Interface (ASAI-BRI) or an Ethernet interface (ASAI-Ethernet).

Intended Audience

This document is written for the application designer responsible for building/programming custom applications and features. This document is also helpful to any individual who needs a functional description of ASAI.

ASAI provides users with the capability to drive a variety of switch features. It is essential, therefore, that readers of this document possess extensive knowledge regarding not only these switch features themselves, but also their interactions.

NOTE:
See the ‘‘Related Documents’’ section for a list of manuals that provide switch feature and ASAI protocol information.

1. In the interest of brevity, CallVisor ASAI is referred to as ASAI throughout the remainder of this manual.

Terminology

Definitions of terms relating to ASAI can be found in the glossary at the end of this document.

Conventions Used in This Document

The majority of conventions in this document are self-explanatory and need not be discussed here. There are, however, some conventions whose meaning may not be obvious to the reader. An explanation of these conventions follows.

Chapters 3 through 11 detail the function of each feature or capability in this version of ASAI. A capability is a request for or an indication of an operation. For example, dropping a party from a call is a capability of ASAI. Related capabilities are grouped into functional sets called capability groups.[2]

Each capability within the group is divided into the following subsections:

- **Capability Name**

  Provides a short overview of the capability and its functions.

- **Information Flow**

  This heading provides information regarding the flow of data from the adjunct to the switch or vice versa. For example, the switch may generate reports to the adjunct (application processor), but the adjunct does not need to respond to these reports. This situation is different when dealing with many of the call control capabilities that require a give and take of data between the switch and the adjunct.

- `<Capability Name>` **Parameter(s)**

  This heading documents the type of information (such as the caller_id) that passes between the switch and the adjunct (usually in the form of a request to the switch). The actual name is based on the capability being discussed; for example, Call Control Parameters.

2. Capability groups are analogous to Application Service Elements (ASEs) described in the *AT&T Adjunct/Switch Application (ASAI) Interface Specification*, 555-025-203.

- **ACK (positive acknowledgement) Parameter(s)**

  There are many instances when the switch simply acknowledges the request made by the adjunct and subsequently performs the operation. There are other times when the switch replies with specific information (such as the identity of the party making the call) to the adjunct within the acknowledgement.

- **Denial (NAK) Cause(s)**

  This heading designates a negative acknowledgement (NAK) from the switch. This means that the information provided by the adjunct to the switch was incorrect; for example, one of the parameters, such as the call_id, was incorrect. At this point the switch rejects the request and terminates the communication channel between the switch and the adjunct. The switch also provides a reason why the operation was not performed. These reasons or causes fall under the Denial (NAK) Cause(s).

  Each ASAI capability contains a Denial (NAK) section with a list of cause calues most commonly occurring. An attempt was made to have these lists complete. However, because of the many unpredictable switch/feature interactions, it is possible that those lists are not complete. Therefore, application and ASAI library writers should be able to handle any other (valid) cause values not listed under the particular capability.

- **Protocol (NAK) Error Cause(s)**

  This heading designates a protocol processing error.

- **Considerations**

  This heading provides the user with any special information that should be taken into account for this particular capability.

Related Documents

AT&T Adjunct/Switch Application Interface Specification (ASAI), 555-025-203

The ASAI Specification document provides a detailed description of the ASAI Reference Model and contains all the capabilities available from ASAI. The *CallVisor ASAI Technical Reference*, on the other hand, describes only those the capabilities available with the Generic 3 switch.

AT&T DEFINITY® Communications System Generic 3 CallVisor ASAI Planning Guide, 555-230-222

This manual provides planning and implementation information for CallVisor ASAI.

AT&T DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference, 555-230-221

The Protocol Reference provides detailed protocol information of CallVisor Adjunct Switch Application Interface (ASAI) for the Generic 3 switch.

NOTE:
Distribution of this document is restricted to AT&T.

AT&T DEFINITY Communications System Generic 3 Installation, Administration, and Maintenance of CallVisor ASAI Over DEFINITY LAN Gateway, 555-230-223

This document describes the installation, administration, and maintenance of the ASAI-Ethernet application, which provides ASAI functionality using 10Base-T Ethernet rather than BRI as a transport media.

AT&T DEFINITY Communications System Generic 3 Feature Description, 555-230-204

The Feature Description serves as an overall reference for the planning, operation, and administration stages of Generic 3 switch.

AT&T DEFINITY Communications System Generic 3 System Description, 555-230-206

This manual provides a technical description of hardware, environmental, and space requirements and parameters.

AT&T DEFINITY Communications System Generic 3 V4 Implementation, 555-230-655

This manual documents the implementation of the Generic 3i switch.

AT&T DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide, 555-230-520

This manual documents call vectoring for the Generic 3 switch.

GBCS Products Security Handbook, 555-025-600

This manual provides information on securing various AT&T products against toll fraud.

NOTE:
With regard to CallVisor ASAI, the importance of security cannot be overestimated. It is just as important to secure the processor the application resides on as it is to secure the PBX.

ASAI and Capability Groups

1

Introduction

The purpose of this chapter is to present an overview of ASAI and the services it provides.

ASAI services are divided into functional sets called capability groups.[1] Capability groups enable the adjunct[2] to communicate with and control the switch.[3]

1. Capability groups are analogous to Application Service Elements (ASEs) described in the *AT&T Adjunct/Switch Application Interface (ASAI) Specification*, 555-025-203.
2. For the purpose of this document, the term *adjunct* is defined as the application processor.
3. ASAI is not limited to a one-to-one correspondence between the switch and an adjunct. Multiple adjunct configurations are available and are discussed in a subsequent section. For the sake of this introduction, however, the scope is limited to a single switch and a single adjunct.

Each capability group is defined by the set of functions within it. ASAI defines eight capability groups in all:

Call Control[4] The capabilities in this group enable the adjunct to place, monitor, and control any party on a single call as it moves through the switch.

Domain (Station/ACD Split) Control[4] The station capabilities in this group enable the adjunct to place, monitor, and control all calls at a specific station domain.

This capability group also enables the adjunct to receive reports as to the status of agents on an ACD split. Currently the switch provides the Logout Event, and, starting with G3V4, the Login Event.

Notification This capability group lets the adjunct request and cancel event reporting on certain calls.

Routing This capability group allows the switch to ask the adjunct for a call's destination. The adjunct supplies the destination based on call-related information (for example, called number).

Request Feature The single capability in this group lets the adjunct request switch features, such as the agent login, logout, work mode changes, Call Forwarding, and Send All Calls (SAC).

Value Query This capability group enables the adjunct to request information regarding switch resources. Using this capability group would, for example, allow a user to query the switch for the number of agents currently logged in to an ACD split.

Set Value This capability group enables the adjunct to set switch-controlled services, such as the Message Waiting Indicator (MWI), for any specified telephone set.

Maintenance This capability group enables the adjunct to suspend and resume switch alarms on the ASAI link. It also enables the adjunct or the switch to request the status of the ASAI software at the remote endpoint using the Heartbeat capability.

4. The Call Control Capability Group is the Single Call Control subset of the Third Party Call Control Capability Group. The Domain Control Capability Group is also a subset of the Third Party Call Control Capability Group, as specified in the *AT&T Adjunct/Switch Interface (ASAI) Specification*, 555-025-203. For the sake of brevity, these groups are referred to as the Call Control Capability Group and the Domain Control Capability Group, respectively. The specific capabilities within each group carry the **Third Party** prefix; for example, Third Party Make Call and Third Party Domain (Station) Control Request.

Capabilities

While capabilities are grouped by the services they may provide, all groups divide their particular capabilities into three categories or types. These categories are: initiating, controlling, and terminating capabilities.

Initiating — These capabilities are used to open a channel of communication between the adjunct and the switch for messaging purposes. An example of an initiating capability is Third Party Make Call that allows the adjunct to direct the switch to place a call.

Controlling — These capabilities are used to exchange information once the channel of communication has been established. For example, Third Party Selective Hold can be used to place a call on hold, or Third Party Merge can be used to transfer a call.

Terminating — These capabilities end or close the channel of communication between the adjunct and the switch. For example, Third Party Call Ended indicates that the call has ended.

Capabilities and Associations

Central to this introduction of capability groups and ASAI in general is the concept of an *association*. An association is defined as a channel of communication between the adjunct and the switch for messaging purposes.

It may be helpful to think of an association as a communications session; each session could involve information pertaining to many calls.

The previous section regarding types of capabilities showed that all capabilities act across a channel of communication which *is* an association. Initiating capabilities begin an association, controlling capabilities manipulate message exchange during the association, and terminating capabilities end the association.

Associations and Capability Groups

ASAI defines eight different types of associations, each of which corresponds to a particular capability group:

- Call Control Associations
- Domain (Station/ACD Split) Control Associations
- Notification Associations
- Routing Associations

- Request Feature Associations
- Value Query Associations
- Set Value Associations
- Maintenance Associations

ASAI and Supported Applications

2

Introduction

This chapter provides a look at the various configurations and applications that can be supported by the ASAI capabilities.

The first part of this chapter presents a simple configuration and several application samples. The latter part provides additional configurations that support the ASAI capabilities, and a table that defines the capacity limitations of ASAI.

Figure 2-1 illustrates the simplest configuration, an adjunct (application processor) connected to a switch via a single ASAI-BRI/Ethernet link.

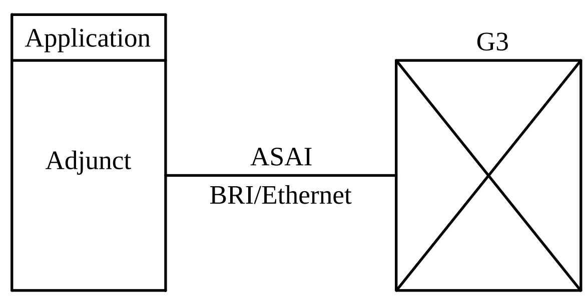

Figure 2-1. Single Link — Single Processor Configuration

The adjunct can be a personal computer, a minicomputer, or a mainframe. Applications on the same adjunct monitor and control voice calls or perform other operations on behalf of a telephone user.

Applications on the adjunct can share the ASAI link when communicating with the switch. The switch does not distinguish between multiple applications that may be sharing a switch link.

A user typically has a telephone and a data terminal at his or her desk. The user can control the voice calls at his or her telephone by using the telephone or entering commands at the data terminal. When using the data terminal, the application controls the voice call via the ASAI link. How the data terminal is connected to the adjunct is irrelevant to the ASAI-supported applications described in this document.

Applications

ASAI supports a variety of application types:

- Those that control a single station (telephone set) on behalf of a specific user
- Those that control all parties on a call
- Those that route incoming calls
- Those that make outbound calls from an ACD split for a telemarketing center
- Those that monitor calls entering vectors and/or ACD splits

The Generic 3 switch allows an application to control a specific extension on a call and, at the same time, allows another application to control another extension on the call. In this case, both applications can independently control endpoints on the call in the same way that users can by using their telephone set.

For G3V3 and later, Multiple Monitors provides the ability for up to three ASAI applications to monitor the same ACD split or VDN domain (instead of just one). In addition, Multiple Adjunct Routing allows link redundancy.

Sample Applications

The sample applications in the following section provide a practical, ‘‘real world’’ illustration of ASAI capabilities.

NOTE:
The applications described in this section are not restricted to any particular configuration described in this section, nor are they mutually exclusive. Any configuration and group of applications can be used simultaneously. The switch does not restrict any mix of applications, except as dictated by capacity and performance constraints. For information on ASAI capacity limits, see Appendix B, ‘‘ASAI and Generic 3 Switch Requirements.’’

In addition, the ASAI interface provided by the switch is not the only system component that might be needed to provide these applications. For example, additional hardware (computer data terminals, voice response units, call classifier boards) and/or software (application interface, call vectoring) might be needed. The ASAI interface only provides the communication link to access the switch services that make these applications possible.

Outbound Call Management

A good example of Outbound Call Management (OCM) is an Outbound Telephone Support Center Application. An Outbound Telephone Support Center Application automatically generates outbound calls that are to be handled by a specified user community (agent pool).

Outbound applications fall into two categories:

Preview Dialing — The agent or user previews a screen of data pertaining to the call and enters information into the system when ready to make the call. Preview dialing allows an agent or user to control when the outbound call is started, enabling the user to prepare for a conversation with the called party.

Predictive Dialing — The adjunct application makes more outbound calls than there are agents. Statistically, a certain number of calls will go unanswered, busy, intercept, or answered by machine detection (if so optioned), etc. The system connects agents only with answered calls. Predictive dialing makes more efficient use of an agent pool by eliminating dialing time, listening to ringing, etc.

The following sample scenarios illustrate the operation of Preview Dialing and Predictive Dialing.

Preview Dialing

1. The agent uses a data terminal to log into the outbound telephone support application. The application establishes a Domain (Station) Control association for the agent. There must be one such association for each agent.
2. The agent enters information indicating readiness to preview data. There must be one such association for each agent.
3. The adjunct application displays a screen of data to the agent.
4. When the agent enters information, the application uses the ASAI Third Party Auto-Dial capability to place an outbound call from the agent's station to the number associated with the displayed data. See ''Domain (Station ACD Split) Control'' in Chapter 5 for more information regarding the Third Party Auto-Dial capability. Alternately, when the agent enters information, the application uses the ASAI Third Party Make Call Capability to place an outbound call from the agent to the number associated with the displayed data. See ''Call Control Capability Group'' in Chapter 4 for more information regarding the Third Party Make Call Capability.
5. The switch sends the adjunct event reports about the call for agent tracking until the call disconnects.
6. The cycle continues.

Predictive Dialing

Predictive dialing uses special hardware, a call classifier. The call classifier is capable of detecting ringing, voice energy, special tones, and an answering machine.

1. A user (agent) uses either a telephone set or data terminal to log into the outbound telephone support application. If the user uses the data terminal, then the adjunct application uses the ASAI Request Feature Capability to log the agent into the ACD split on the switch.
2. The application uses the Third Party Make Call Capability with Service Circuit/Call Classifier and Alert Destination First options to make outbound calls from the ACD split extension to external numbers. This is called a ''switch-classified'' call. Typically, these numbers come from a calling list maintained for the outbound telemarketing application. The application uses queries to monitor switch resources such as agents logged into the split, available classifiers, and available trunk resources. The application usually has a pacing algorithm that places calls ahead of available agents.
3. When the call classifier detects answer or an answering machine, the switch ACD software distributes the call to an available agent or queues the call if no agent is available. The switch software can be configured to drop calls if an answering machine is detected (if AMD is in use).

4. The switch provides the adjunct application(s) with event reports for call activity within the ACD split. The application, in turn, might display information from the calling list to an agent when the switch ACD software connects an outbound call to an agent.
5. The cycle continues.
6. For G3V3 and later, the Answering Machine Detection feature may be used in conjunction with this type of dialing to receive Connected Event Reports on any type of trunk.

Inbound Call Management

Inbound Call Management (ICM) provides inbound telemarketing centers with the ability to increase ACD agent efficiency and tracking by enabling ICM applications to:

- Monitor (receive ASAI Event Reports) all calls delivered to Vector Directory Numbers (VDNs) and ACD splits and calls originated by ACD agents or users
- Route calls to specific ACD/hunt groups, VDNs, or ACD agents based on incoming call information [for example, Calling Party Number/Billing Number (CPN/BN)[1] and Dialed Number Identification Service (DNIS)][2] and ACD call activity (for example, total number of calls queued, or number of available agents)
- Prepare and deliver, together with the voice call, the appropriate data screen to the selected agent or user
- Duplicate and transfer the caller's data screen when an ACD agent or user conferences or transfers the voice call to another destination (for example, ACD supervisor, or expert agent)
- Provide ACD agent functions (for example, login, logout, or work mode change) from a data terminal
- Provide ACD agents and/or supervisors with Internally Measured Data on the performance of agents, splits, trunk groups, or VDNs.
- Set the billing rate for calls to a 900-number with AT&T MultiQuest Vari-A-Bill service.

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1. CPN/BN information can be used by the ICM application to identify the caller so that the caller information can be retrieved from an application database. In addition, CPN/BN allows the application to gather statistics about the callers' geographical locations and to assess the effectiveness of different marketing campaigns.
2. DNIS can be used by the ICM application to identify the type of service or product the caller is calling about. This allows a single agent to handle multiple services or products without asking the caller for the service requested. For example, a single agent could handle questions about Product A and Product B by assigning each product a different telephone number. When a call is delivered to the agent, the application, based on the DNIS received, displays the appropriate product information that allows the agent to service the caller for his or her specific need.

The following sample scenarios illustrate the operation of several ICM applications.

ACD Call Activity Monitoring

The ACD Call Activity Monitoring Application uses event reporting to track the call activity of VDNs, ACD splits, and individual agents or users. (For G3V3 and later, Multiple Monitors provides the ability for up to three ASAI applications to monitor the same ACD Split or VDN domain.)

The application may use the event reports to generate ACD reports containing information such as the following:

- Call distribution by CPN/BN[3] for each DNIS
- Total number of calls handled by each VDN, ACD split, and/or agent
- Total number of calls, with CPN/BN, that abandon/drop while in queue
- Total number of ACD, agent-to-agent, agent-to-supervisor, and personal calls that were originated and received by each agent
- Average and maximum time in queue
- Average and maximum queue length
- Average and maximum call holding time
- Average time spent by each agent on a call
- Total number of calls that interflow/intraflow

In addition, if the application has complete control of the agent work modes, such as in adjunct-controlled splits, the agent activity reports can also be generated.

A sample scenario for the ACD Call Activity Monitoring application is as follows:

1. The application uses the Event Notification Request Capabilities and Domain Control Capabilities to monitor all calls delivered to ACD splits and all calls originated and delivered to an agent station.
2. The switch sends event reports (for example, Call Initiated, Alerting, Connected, Call Transferred, or Dropped) to the application for all calls entering the monitored ACD splits and stations.
3. The event reports allow the application to produce the ACD Call Activity reports described previously.

The application can also make use of Internally Measured Data (stored by the switch) on which VuStats is based.

3. For more information on CPN/BN and DNIS administration, refer to *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test*, *Issue 4*, 555-230-104 and *AT&T DEFINITY Communications System Generic 3 Implementation*, *Issue 1*, 555-230-653.

Data Screen Delivery and Voice/Data Transfer

A Data Screen Delivery and Voice/Data Transfer application may use CPN/BN or calling party number, DNIS or called party number, and answering destination information to construct and deliver a data screen to the answering agent/user's data terminal. Likewise, when an agent or user conferences or transfers a call, the application uses the conferenced agent or transferred-to agent information to automatically transfer the data screen to the new agent handling the call.

A sample scenario for the Data Screen Delivery and Voice/Data Transfer application is as follows:

1. The application uses the Event Notification Request capability to monitor all incoming calls to an ACD split or VDN.
2. When a call enters the monitored ACD split or VDN, the switch sends a Call Offered to Domain Event Report containing the call's CPN/BN, DNIS, UUI (whether supplied by the network or by some other CallVisor adjunct), and any lookahead interflow and collected digits information associated with the call.
3. The application does a database search on the caller information (CPN/BN) and retrieves the caller's data to fill a data screen based on the service dialed (DNIS).
4. When the call is delivered to an available agent and/or user, the switch sends an Alerting and/or Connected Event Report containing the number of the agent or user handling the call. The application then delivers the assembled data screen to the data terminal associated with the agent or user handling the call.
5. If the agent or user conferences or transfers the call to another destination, the switch sends a Call Conferenced or Call Transferred Event Report indicating the new destination. The application then duplicates or recreates the caller's data screen at the data terminal associated with the new destination.
6. When an agent or caller disconnects and/or drops, the Disconnect/Dropped Event Report is generated and the application may take appropriate clean up actions.

Data Screen Delivery with Call Prompting

The application can also use the switch-based Call Prompting feature to obtain additional information (for example, account number) from the caller. The entered information can be used to select the appropriate data screen.

A sample scenario for the Data Screen Delivery with Call Prompting application is as follows:

1. The customer administers a vector with a Collect Digits command as part of the Call Prompting feature.
2. The application uses the Event Notification Request capability to monitor all incoming calls to the VDN used to distribute calls to agents or users.
3. When a call enters the monitored VDN, the switch sends a Call Offered to Domain Event Report containing the digits collected for the call in the Collect Digits vector command.
4. The application does a database search on the digits collected and retrieves the caller's data to fill a data screen based on the service dialed.
5. When the call is delivered to an available agent and/or user, the switch sends an Alerting and/or Connected Event Report containing the number of the agent or user handling the call. The application then delivers the assembled data screen to the data terminal associated with the agent or user handling the call.

Speech Processing Integration

Speech Processing Integration can be achieved if the application uses a Voice Response Unit (VRU) to interact with the caller. The VRU is an adjunct and calls are delivered to VRU ports for announcements and collection of additional information from the caller. The application communicates with the VRU software and uses the information provided by the caller to prepare the appropriate data screen and/or route the call to the appropriate destination (for example, ACD agent).

A sample scenario for the Speech Processing Integration application is as follows:

1. The customer administers the VRU ports as ACD agents in an ACD split. To the switch the VRU ports look like ACD agents.
2. The application uses the Event Notification Request capability to monitor all incoming calls to the ACD split associated with the VRU ports.
3. The application uses the Request Feature capability to log in, log out, and change work modes of the VRU ports. It is recommended that auto-available agents be used for VRUs so that this Request Feature capability does not have to be invoked.

4. When a call enters the monitored ACD split or VDN, the switch starts sending Event Reports to the application about the call including the Call Offered to Domain Event Report containing the call's CPN/BN, DNIS, and any lookahead interflow and collected digits information associated with the call.
5. When a call is connected to the VRU, the application uses the VRU's voice processing capabilities to interact with the caller. The caller, after interacting with the VRU (for example, listening to account balances or transferring funds), may choose to talk to an agent.
6. The application uses Call Control Capabilities (for example, Third Party Selective Hold, Third Party Make Call, and Third Party Merge) to transfer the call to the agent or group of agents (ACD split) designated to handle this type of caller.
7. When the call is delivered to an available agent, the switch sends an Alerting and/or Connected Event Report containing the number of the agent handling the call. The application then delivers the assembled data screen to the data terminal associated with the agent handling the call. Typically, the VRU is handing this information off to a host that will be delivering the data screen to the appropriate agent.

Adjunct Routing

The Adjunct Routing application allows the switch to request call routing instructions (that is, destination) from an adjunct application.

A sample scenario for the Adjunct Routing application is as follows:

1. The customer administers a vector with an Adjunct Routing command as part of the Call Vectoring feature.
2. When vector processing for a call encounters an adjunct routing vector command, the switch sends a Route Request Capability requesting a route for the call. The route request includes the call's CPN/BN and VDN number used by the call to access the vector.
3. The application selects the route for the call based on the call information passed and/or agent availability and sends a Route Select Capability with the route (that is, internal or external telephone number). The switch then routes the call as indicated by the application.
4. When the call is delivered to the destination, the switch sends a Route End Capability indicating a successful route. If the call cannot be routed to the specified destination, the reason for failure (for example, destination busy or invalid number) is returned to the application.

The Call Prompting feature can also be used to collect additional information from the caller before the switch requests a route from the application.[4] The application then uses the collected digits (for example, Sales, Parts, or Service department selection) to select the appropriate route for the call.

For G3V3 and later, the following features provide additional functionality:

ASAI-Provided Digits allows an adjunct to include digits in a Route Select capability. These digits are treated as dial-ahead digits for the call. They can be collected one at a time or in groups using the **collect digits** vector command(s).

For example, an incoming call is routed by an application to a VRU. As part of the ‘‘route,’’ the application provides digits to be passed to the VRU. The digits may indicate which application or script to play, which host is handling the call (in case there are multiple hosts), and perhaps a call identifier. These digits can be collected during vector processing through a ‘‘"collect’’ step and then passed to a VRU via the ‘‘converse’’ step, or to another application via the route request or Call Offered to Domain event report.

ASAI-Requested Digit Collection gives an adjunct the ability to request that a DTMF tone detector (TN744) be connected to detect user-entered digits. The request is made via an option of the **Route Select** message. The digits collected as a result of this feature are passed to ASAI monitoring and/or controlling adjuncts for action. The switch handles these digits like dial-ahead digits.

These digits are collected while the call is not in vector processing. They are sent to an ASAI adjunct, and/or they may be used by Call Prompting features.

Multiple Outstanding Route Requests allows multiple ASAI Route Requests for the same call to be active at the same time. The Route Requests can be over the same or different ASAI links. The requests are all made from the same vector. This feature is used for load balancing among multiple applications or to implement redundancy in adjunct routing.

User to User Information (UUI) allows distributed CallVisor ASAI and ACD users to associate caller information with a call. This information may be a customer number, credit card number, alphanumeric digits, or a binary string. It is propagated with the call whether the call is transferred or routed to a destination on the local switch or to a destination on a remote switch.

Information may be sent by the network or by ASAI. It is stored with calls and passed to the network and to ASAI.

4. The last set of digits collected is sent to the application in the switch request for a route.

For G3V4 and later, the following features provide additional functionality:

Flexible Billing: The adjunct will be informed if Flexible Billing is available on an incoming call. This information could be used to route calls without this ability differently.

Logging for Call Back

The Logging for Call Back application uses CPN/BN or calling party number and any digits collected via the Call Prompting Feature to record the caller's phone number and allow the caller, who otherwise might wait in queue for an extended period, to disconnect from the call. The application will then call the disconnected caller when agents are available to handle the call.

A sample scenario for the Logging for Call Back application is as follows:

1. The application uses the Event Notification Request capability to monitor all incoming calls to an ACD split or VDN.
2. During periods of high call activity with many queued calls, the caller receives an announcement with the following options: to leave a phone number where the caller can be reached, to drop from the call (if the CPN/BN received is recognized), or to wait in queue.
3. After the caller provides the phone number, the PBX sends the call information (for example, CPN/BN, DNIS, collected digits) to the application in a Call Offered to Domain Event Report or a Route capability and disconnects the call.
4. When agents are available, the application uses the Outbound Telephone Support application (for example, Predictive Dialing) to return calls to the disconnected callers.

Automatic Agent Reconfiguration

The Automatic Agent Reconfiguration application uses the Request Feature Capability to move agents (that is, login and logout) to different ACD splits based on the call activity levels (for example, queue length, time in queue) of the splits. The application increases the number of agents available to handle the queued calls by moving an agent from other ACD splits that can be staffed with fewer agents.

A sample scenario for the Automatic Agent Reconfiguration application is as follows:

1. The application uses the Event Notification Request capability to monitor several ACD splits.
2. The application tracks the number of calls in the queue and the number of available agents for each ACD split. The switch sends the Queued Event Report every time a new call queues into an ACD split. The Value Query Capability provides the number of available agents for each ACD split.

3. Based on application-specific thresholds (for example, number of calls in queue) the application uses the Request Feature Capability to log in an agent into another split.

Sequence Dialing

Sequence Dialing is implemented in two ways:

1. By using VDN Return Destination (release G3V2 and later), in which callers reach a VDN with Sequence Dialing activated. Through the VDN Return function, they reach a final destination but do not hang up when the other side drops. This will automatically return them to VDN processing and give them the ability to call other numbers.
2. By using ASAI-Requested Digit Collection in conjunction with an ASAI application. (This is a combination of ASAI and VDN Return Destination.) ASAI collects a certain digit sequence that indicates sequence dialing is desired. The caller reaches a VDN, is transferred to a final destination that results in a busy or unanswered call, and enters a specific sequence, such as a ‘‘#’’, to make another call. The application receives the digits through an Entered Digits Event Report. It then drops the far end and returns the call to VDN Return Destination for repeat dialing.

Office Automation

Office Automation applications allow office personnel (users) to use the computer data terminal to logically integrate voice and data handling at the user’s desktop by allowing an application to:

- Know the status of calls at the user’s phone
- Initiate, terminate, and control (hold, reconnect, transfer, conference) calls at the user’s phone
- Invoke switch features (that is, Call Forwarding, Send All Calls [SAC]) on behalf of the phone
- Provide messaging services that integrate the Message Waiting Indicator at the user’s phone

Incoming Call Identification

The Incoming Call Identification (ICI) application displays the calling party name and telephone number on the user’s data terminal. Based on the displayed information, the user can decide whether to answer the call or send the call to coverage.

A sample scenario for the Incoming Call Identification application is as follows:

1. The application uses the Domain Control capability to monitor all calls originated and received by or delivered to users' phone sets.
2. When an incoming call is delivered to a user's phone, the switch sends an Alerting Event Report containing the call's CPN/BN and dialed number (implies redirection if different from the alerting phone).
3. The application displays, at the user's data terminal, the information contained in the Event Report as the call rings the user's phone. When CPN/BN is available, the application searches its own database[5] (for example, corporate directory or customers' database) of names and telephone numbers, so that the calling party name is also displayed on the user's terminal.
4. Based on the information displayed at the data terminal, the user can answer the call or invoke the SAC feature from either the data terminal or the telephone set. Alternately, the application may request Redirect Call based on the information obtained.

Phone Management and Directory Services

A Phone Management and Directory Services application may allow telephone users to:

- Originate, answer, and manipulate calls at a station by using hold, transfer, reconnect, answer, conference, and drop
- Request to make a call using the called party name
- Create a personal directory list (the user might define his or her own directory to be used by the application when searching for a telephone number to be included in call request)
- Redirect calls to the message desk or coverage

A sample scenario for the Phone Management and Directory Services application is as follows:

1. The application uses the Domain (Station) Control capability to monitor all calls at a station.
2. A user brings up the telephone management screen at his or her data terminal and enters the name of the called person.
3. The application searches the user's personal directory and corporate directory for the phone number associated with the called name. As soon as a telephone number is retrieved, the application uses the Third Party Auto Dial capability to originate the call for the user.

5. The user may define the personal directory database to be used by the application.

4. The application receives event reports (for example, Call Originated, Alerting, Connected, Call Conferenced, Dropped) for the call indicating the status and/or progress of the call. The application will then display the status of the call at the user's data terminal.
5. The user controls the call by entering telephone commands at the data terminal. The application then uses the Call Control capabilities to control (for example, hold, transfer, conference) a call at the user's station.
6. The user requests the application to forward or redirect calls to the message desk or coverage. The application uses the Request Feature capability to request SAC or Call Forwarding on behalf of the station associated with the user. Alternately, the application may request Redirect Call based on the information obtained.

Message Desk

A Message Desk application may provide users with dialing and messaging services. These services may allow office personnel to take messages from callers, look up numbers in an electronic directory, and use on-screen commands to make, receive, and manipulate telephone calls (for example, hold, transfer). In addition, the Message Desk application may control the state of the message waiting lamp to notify telephone users that voice and/or text mail, as well as other messages, are waiting for the user.

A sample scenario for the Message Desk application is as follows:

1. The application uses the Domain (Station) Control capability to monitor the group of stations designated as the message desk stations (for example, secretary, coverage point).
2. When a call is redirected to the message desk (via Send All Calls or call coverage), the application receives an Alerting Event Report containing the original dialed number, the calling party number, the alerting station number, and, if link version 2 is active, the reason for redirection.
3. The application uses the dialed, calling party, and alerting station numbers to search and automatically display on the message desk data terminal the messages left by the originally called party for the caller. The called party has used electronic mail to generate and send to the message desk application the messages that should be provided to callers by the message desk attendant.[6] If no message is provided, a standard message is given to the caller.

6. The user might provide one standard message to all callers (for example, busy in a meeting all day) or different messages to different callers based on the calling party number (for example, off-PBX calls receive: busy in a meeting; on-PBX calls receive: busy in a meeting with John Doe until 5:00 p.m.).

4. Messages left at the message desk are sent via electronic mail to the originally called person with the calling party number automatically added to the electronic mail message. In addition, the application uses the Set Value capability to turn on the message waiting lamp at the voice set.
5. After the user has read the messages left at the message desk, the application uses the Set Value capability to turn off the message waiting lamp at the voice set.
6. If the application also provides voice mailboxes, the application can allow the user to listen to voice mail messages by using the Third Party Auto Dial capability to set up a call between the user and the user's mailbox. The user then uses the data terminal to listen, delete, forward, annotate, skip, and save the voice mail messages.
7. The user can also request to make a call based on the telephone number or calling party name provided in the message center electronic mail message. The application uses the Third Party Auto Dial capability to originate the call while the user continues reading his or her messages.

Additional Configurations

Figures 2-2 to 2-4 show additional ASAI configurations supported in Generic 3. The applications supported for these configurations are the same as those previously described in this section.

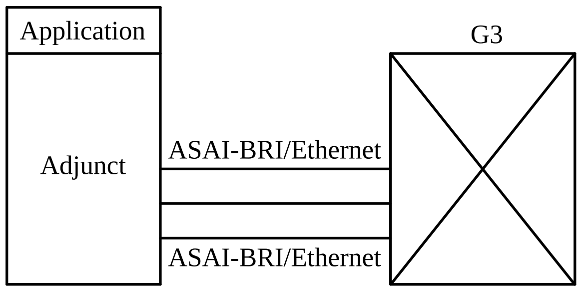

Figure 2-2. Multiple Link — Single Processor Configuration

Figure 2-2 shows a single adjunct connected to the switch via multiple ASAI links. The multiple links may be used to support multiple applications on the same adjunct. This does not mean, however, that one link per application is required. Multiple applications can run on a single link. From the switch's point of view, each link is a single application and no correlation is made between link associations or applications. The switch does not provide any automated link backup procedures.

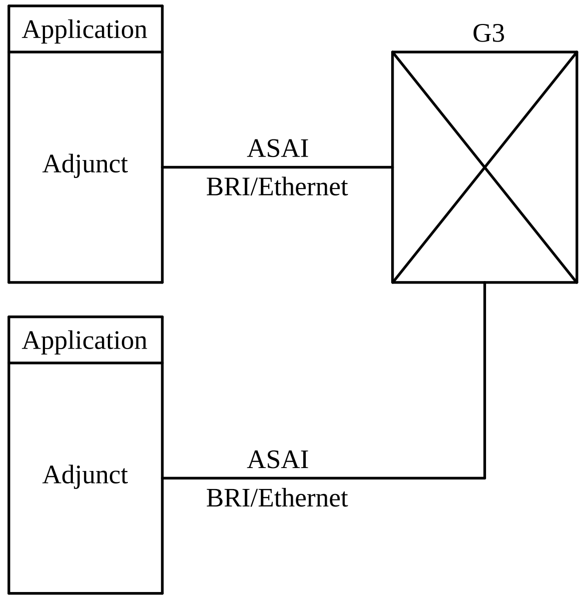

Figure 2-3. Single Link — Multiple Processors Configuration

Figure 2-3 shows multiple adjuncts using their own ASAI links to communicate with the switch. Customers who have separate applications for different telemarketing groups or who provide telemarketing and office automation functions for groups on the same switch may use this configuration. From the switch's point of view, this configuration looks the same as the one in Figure 2-2.

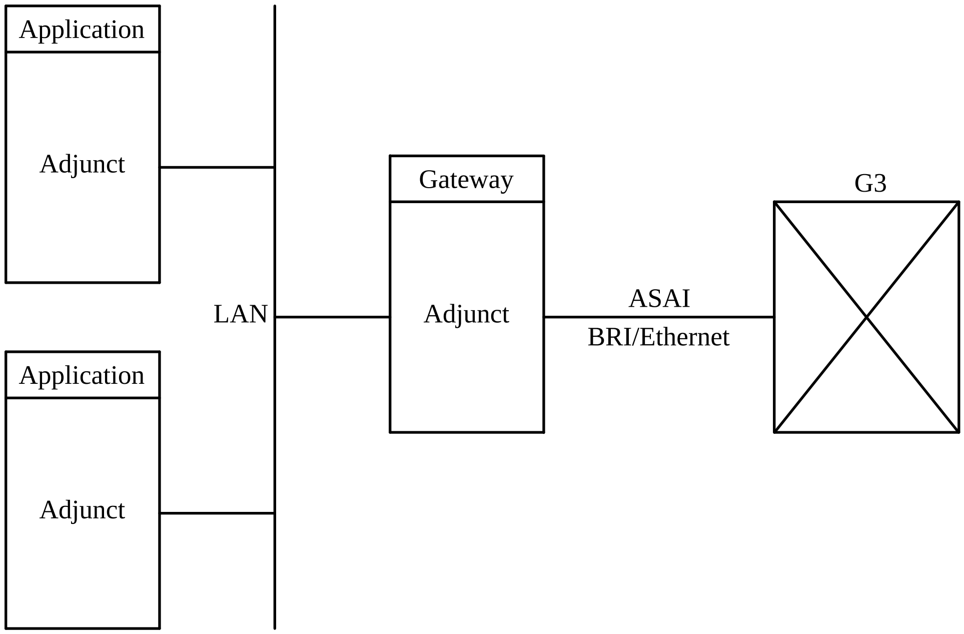

Figure 2-4. Single Link — Gateway/Server Configuration

Figure 2-4 shows multiple adjuncts communicating with a Generic 3 switch via a single ASAI link. Each adjunct is independent of each other and the ASAI link is managed by a single adjunct. This adjunct serves as the ‘‘gateway’’ between the data Local Area Network (LAN) environment and protocols and the voice environment. The gateway adjunct can manage more than one ASAI link depending on the ASAI traffic generated by the adjunct processes. From the switch’s point of view, the ASAI link is a single application.

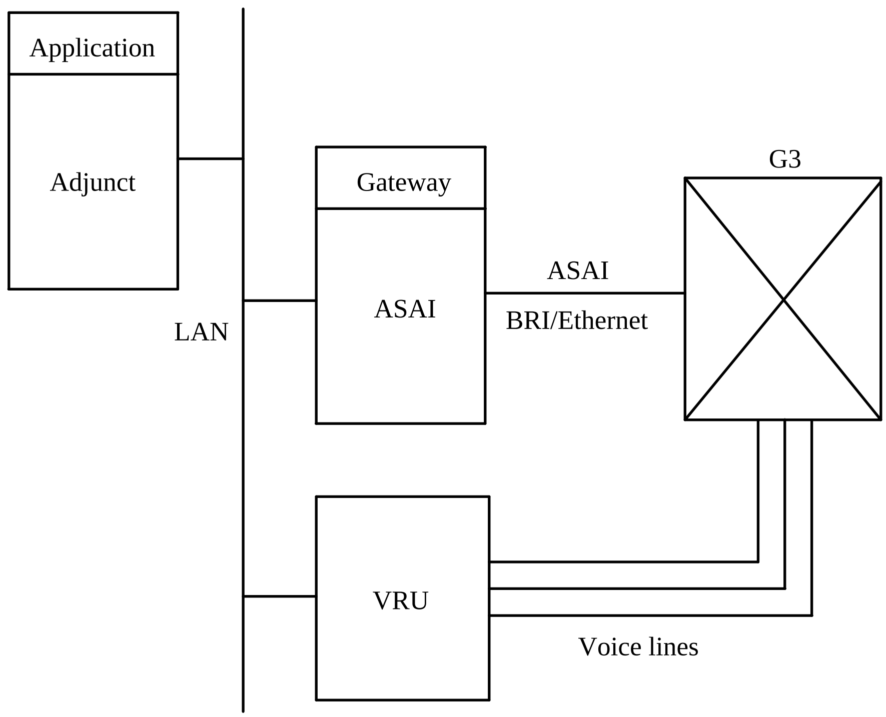

Figure 2-5. ASAI Integration with a VRU Configuration

In Figure 2-5 the application shown uses ASAI together with voice response services to control calls. For example, incoming calls might terminate on the VRU where VRU software collects additional information. Using this information the application might then make an ASAI request to transfer the call to its final destination.

NOTE:
Users should work with their AT&T account team to carefully evaluate the impact of ASAI applications upon the switch processor. This impact is a function of the specific ASAI application and its interaction with switch features such as Basic Call Vectoring. It is recommended that account teams contact AT&T for assistance in evaluating such impact. Call the AT&T Design Center at 1 800 521-7872. For existing applications, call the TSC at 1 800 344-9670.

Event Reporting and U-Abort Capabilities

3

Introduction

Common capabilities are those capabilities used by more than one capability group. There are two common capabilities: Event Reporting and U-Abort (User Abort).

This chapter describes these capabilities and the capability groups where they are present.

Event Reporting Capability

This capability is used by the switch to send call-related information to an adjunct. For example, if a user makes a call from his or her telephone, event reports are sent to an adjunct regarding the call, provided that event reporting has been requested for this particular telephone or call.

NOTE:
Event reporting does not generate screen-formatted or hardcopy reports. Event reporting, for the purposes of ASAI, simply means that call information is provided to an adjunct by the switch. Event Reports are ‘‘informational only.’’ To control a call, an application must use call control capabilities.

U-Abort Capability

This capability notifies either the switch or the adjunct that processing for the association is ending abnormally. For example, when switch resources are not available to place a call, as requested by the adjunct, a U-Abort is returned.

Event Reports

Capability Groups and Event Reporting

The Event Reporting capability can be invoked from the following capability groups:

- Call Control Capability Group
- Domain (Station or ACD Split) Control Capability Group
- Notification Capability Group

See Chapters 4, 5, and 6, respectively, for discussions of these capability groups.

An administrable option called ‘‘Event Minimization’’ is available for each ASAI link. This option may be used when event reports would normally be sent on multiple associations, but the adjunct does not need to see more than one. Typically, these event reports are identical except for the association they are sent over (for example, call control, domain control, active notification). Some applications discard duplicate events, so in this case, there is no point in sending them across the ASAI link. When enabled, this option allows only a single such event to be sent. The selection of the association on which the event will be sent is based on association precedence as follows: active notification (if enabled), call control (if enabled), domain control (if enabled).

The Station form is used to change this option. The new option settings take effect the next time the ASAI link is activated.

Call-Related Event Reporting

The adjunct receives call-related event reports from the switch for the following call types:

- Controlled Calls — Calls controlled by the adjunct via the Call Control Capability Group
- Domain-Controlled Calls — Calls controlled by the adjunct via the Domain (in other words, station) Call Control Capability Group
- Monitored Calls — Calls for which the adjunct has requested event reports via the Notification Capability Group

NOTE:
When calls are controlled in some way by the first two capability groups listed above, event reports are provided to the adjunct. Call monitoring (Event Notification) **does not** have to be invoked separately via the Notification Capability Group in order for event reports to be generated.

Non-Call-Related Event Reporting

The event reports mentioned previously were all related to a specific call. An event report that is not directly related to a specific call can also be generated. When an agent logs out of a split/skill that is under Domain (ACD split) Control, a Logout Event Report is generated. Starting with G3V4, a Login Event Report is generated when an agent logs into this type of split. These reports are discussed in detail later in this section.

Information Flow

The adjunct does not respond to event reports. (An adjunct is not required to send a response when an Event Report is received.)

Parameters

event_name [mandatory] Specifies the event being reported

item_value_list [optional] Contains a list of items and their values for the event being reported

Event Reports and Corresponding Items

Every event report issued by the Generic 3 switch contains pieces of information that, individually, are called **items**. Item combinations contain information about the specific event being reported. The following table presents the different event reports and the corresponding items available to the Event Reporting capability.

In general, event reports are not sent for split or vector announcements or attendant group 0.

Event Report	Items Provided with Each Event Report
Alerting	■ Calling Party number (CPN/BN) or ■ Trunk Group number and Trunk Group Member number — only provided if the Calling Party number is unavailable ■ Called Party number (originally dialed number) ■ Cause ■ Connected Party number (alerting number) ■ Call_id ■ Party_id

Event Report	Items Provided with Each Event Report
Alerting, cont'd.	■ Domain (ACD split associated with the call — if any) ■ User to User Information (UUI) ■ Reason for Redirection
Answered	■ Called Party number ■ Connected Party number (answering party number) ■ Call_id ■ Party_id ■ Cause (Special Information Tone [SIT] or Answering Machine Detection [AMD] — if any)
Busy/Unavailable	■ Called Party number (busy number) ■ Call_id ■ Cause
Call Conferenced	■ Calling Party number (conference initiator party number) ■ Called Party Number (newly added party number) ■ Other Call_id ■ Resulting Call_id ■ Party_id List (up to six party_ids) ■ Extension List (up to six extensions)
Call Ended	■ Call_id ■ Cause
Call Initiated	■ Call_id ■ Party_id
Call Offered to Domain	■ Calling Party number (CPN/BN) or ■ Trunk Group number and Trunk Group Member number — only provided if the Calling Party number is unavailable

Event Report	Items Provided with Each Event Report
Call Offered to Domain, cont'd.	■ Called Party number (DNIS) ■ Call_id ■ Item indicator (User-Entered Information — contains up to 16 digits from the most recent Collect Digits vector step) ■ Lookahead Interflow Information ■ Domain (ACD split or VDN) ■ User to User Information (UUI) ■ Flexible Billing
Call Originated	■ Call_id ■ Party_id ■ Calling Party number ■ Called Party number ■ Party extension
Call Redirected	■ Call_id
Call Transferred	■ Calling Party Number (transfer initiator party number) ■ Called Party Number (transferred party number) ■ Other Call_id ■ Resulting Call_id ■ Party_id List (up to six party_ids) ■ Extension List (up to six extensions)
Connected	■ Calling Party number (CPN/BN) or ■ Trunk Group number and Trunk Group Member number — only provided if the Calling Party number is unavailable ■ Called Party number (DNIS) ■ Connected Party number ■ Call_id

Event Report	Items Provided with Each Event Report
Connected, cont'd.	■ Party_id ■ Cause
Cut-Through	■ Call_id ■ Party_id ■ Progress Indicator
Disconnect/Drop	■ Connected Party number (dropped number) ■ Call_id ■ Party_id ■ Cause ■ User to User Information (UUI)
Entered Digits	■ Digits
Hold	■ Connected Party number (number that placed the call on hold) ■ Call_id ■ Party_id
Login	■ Work Mode ■ Agent Physical Extension ■ Agent Logical Extension
Logout	■ Agent Physical Extension ■ Agent Logical Extension
Queued	■ Called Party number (DNIS) ■ Call_id ■ Calls in Queue ■ Domain (ACD split)

Event Report	Items Provided with Each Event Report
Reconnected	■ Connected Party number (number that reconnected to the call) ■ Call_id ■ Party_id
Reorder/Denial	■ Called Party number (default unknown) ■ Call_id ■ Cause
Trunk Seized	■ Called Party Number ■ Call_id ■ Party_id

Alerting Event Report

The switch sends the Alerting Event Report for monitored, controlled, and domain-controlled calls when the following events occur:

- The originator of a switch-classified call[1] is an on-PBX station, and ringing or zip tone is started.
- When a call is redirected to an off-PBX station and the ISDN ALERTing message is received from an ISDN-PRI facility.
- When a switch-classified call is trying to reach an off-PBX station, and the call classifier detects either precise, imprecise, or special ringing.
- When a switch-classified or user-classified call is placed to an off-PBX station, and the ALERTing message is received from the ISDN-PRI facility.

NOTE:
When a classifier and an ISDN-PRI facility both report alerting on a switch-classified call, then the first occurrence generates an Alerting Event Report; succeeding reports are not reported by the switch.

Consecutive Alerting Event Reports are possible in the following cases:

- A station is alerted first and the call goes to coverage; an Alerting Event Report is generated each time a new station is alerted.
- A principal and its bridging users are alerted; an Alerting Event Report is generated for the principal and for each bridged station alerted.
- A call is alerting a Terminating Extension Group (TEG); one report is sent for each TEG member alerted.
- A call is alerting a Personal Central Office Line (PCOL); one report is sent for each PCOL member alerted.
- A call is alerting a coverage/answer point; one report is sent for each alerting member of the coverage answer group.
- A call is alerting a principal with SAC active; one report is sent for the principal and one or more are sent for the coverage points.

The Alerting Event Report is *not sent* for calls that connect to announcements as a result of ACD split forced announcement or announcement vector commands.

When an already queued ASAI-monitored call reaches a converse vector step, the ALERTing message sent to the ASAI host will include an optional cause (CS3/23) to inform the ASAI host that the call has not been de-queued.

1. Switch-classified and user-classified calls are a type of Third Party Make Call. See Chapter 4, ‘‘Call Control Capability Group’’ for a detailed discussion.

Report Items

The following is a list of items provided with this report:

Calling Party number	■ For outgoing calls over PRI facilities — ‘‘calling number’’ from the ISDN SETUP message. ■ For outgoing calls over non-PRI facilities or on-PBX calls — locally originating extension. ■ For incoming call over PRI facilities — ‘‘calling number’’ from the ISDN SETUP message. ■ For incoming calls over non-PRI facilities, the calling party number is generally *not* provided. In this case, the Trunk Group number is provided instead. ■ For calls originated at a bridged call appearance — the principal’s extension. ■ For incoming DCS calls, if the DCS calling party information is available to the switch (if a station with a display gets it), this information is also made available to ASAI. Otherwise, the calling party information is provided as the default.
Trunk Group number/Trunk Group Member number	■ The Trunk Group number and Trunk Group Member number are only provided if the Calling Party number is unavailable.
Called Party number (DNIS)	■ For outgoing calls over PRI facilities, the called number as in the SETUP message. ■ For outgoing calls over non-PRI facilities, the called number is the default trunk value (#####). ■ For incoming calls over PRI facilities, the called number is the one provided in the SETUP message. ■ For incoming calls over PRI facilities to a VDN that does lookahead interflow on calls, if the lookahead interflow attempt fails, the called number provided is the principal extension of the dialed number. If the interflow attempt is successful, the Called Party number is provided as the default. ■ For incoming calls over non-PRI facilities, the called number is the principal extension (may be a group [TEG, hunt group, VDN] extension)[2]

2. If the switch is administered to modify the DNIS digits, then the modified DNIS string is passed.

- For incoming calls to PCOL, the called number is the default extension value (*****).
- For incoming calls to a TEG (principal) group, the TEG group extension is provided.
- For incoming calls to a principal with bridges, the principal's extension is provided.

Connected Party number (alerting party number)

- For outgoing calls over PRI or non-PRI facilities — the default trunk value (#####)
- For incoming calls — locally alerting extension (primary extension for TEGs, PCOLs, bridging). If the party being alerted is on the switch, then the extension of the party being alerted is passed.

Call_id This switch-assigned call identifier is used to associate event reports to calls and to identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

User to User Information If UUI is stored with the call, then that UUI is included in the Alerting Event Report. This applies to UUI that originated in an ISDN PRI/BRI SETUP message; in an ISDN DISCONNECT message; or in an ASAI Route Select, 3rd Party Make Call, or 3rd Party Auto-Dial message.

Reason for Redirection The reason the call has been presented and is alerting at a domain-controlled station. Only applies to calls redirected by the switch or by ASAI from the original destination.

NOTE:
This event report is *not* guaranteed for each call, such as an outbound call that is not end-to-end ISDN, or in cases where you connect a party on an existing call by bridging in or by using the pick up feature, for example. In both cases, the application would see the Connect Event Report without seeing the Alert.

Answered Event Report

The Answered Event Report is only sent for the destination of a switch-classified call, as follows:

- When a switch-classified call is placed to an on-PBX station and the station has answered the call (picked up handset or connected after zip tone).
- When a switch-classified call is placed to an off-PBX destination and an ISDN CONNect message is received from an ISDN-PRI facility.
- When a switch-classified call is placed to an off-PBX destination and the call classifier detects an answer, an answering machine, or a Special Information Tone (SIT) administered to answer. For specific SIT and AMD values, refer to Table 4-2, Detected SITs, in Chapter 4, ''ASAI and Call Control.''

NOTE:
Only one Answered Event Report is possible for a call.

Report Items

The following is a list of items provided with this report:

Called Party number

- For outgoing calls over PRI facilities, the called number as in the SETUP message
- For outgoing calls over non-PRI facilities, the called number is the default trunk value (#####)
- If the destination is on the local PBX, then the extension of the destination called is passed

Connected Party number (answering party number)

- For calls over PRI facilities — ''connected number'' from the PRI CONNect message
- For calls over non-PRI facilities — the default trunk value (#####)
- For calls answered by an on-PBX extension, this is the answering extension.

Call_id — This switch-assigned call identifier is used to associate event reports to calls and to identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id — The switch identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call identifier.

Cause Contains Special Information Tone (SIT) and Answering Machine Detection (AMD) values (if any). For specific SIT and AMD values, refer to Table 4-2, Detected SITs, in Chapter 4, ''ASAI and Call Control.''

Busy/Unavailable Event Report

The Busy/Unavailable Event Report is sent when the destination of a call is busy, as follows:

- When a call is delivered to an on-PBX station and the station is busy (without coverage and call waiting)
- When a call tries to terminate on an ACD split that is not vector-controlled, and the destination ACD split's queue is full, or when no agents are logged in or all agents are in AUX mode, and the ACD split does not have coverage
- When a call encounters a busy vector command in vector processing
- When a Direct-Agent call tries to terminate an on-PBX ACD agent and the specified ACD agent's split queue is full and the specified ACD agent does not have coverage
- When a call is trying to reach an off-PBX party and an ISDN DISConnect message with a User Busy (CS0/17) cause is received from an ISDN-PRI facility
- When a call enters a split for which there are no logged-in agents (CS0/17)
- When a trunk is maintenance or administration busy, or the administration on two sides of the trunk is not the same (CS0/6)
- When an adjunct is attempting to make a call using ARS and there are no available trunks with appropriate bearer capability class (CS0/58).

The Busy/Unavailable Event Report is *not* sent under the following circumstances:

For a switch-classified call, when any of the above conditions occurs, the Third Party Call Ended capability is sent to the adjunct to indicate that the call has been terminated. The call is terminated because a connection could not be established to the destination.

Report Items

The following is a list of items provided with this report:

Called Party Number (DNIS)

- For outgoing calls over PRI facilities, the called number as in the SETUP message
- For outgoing calls over non-PRI facilities, the called number is the default trunk value (#####).
- For calls to a TEG (principal) group, the TEG group extension is provided.
- For calls to on-PBX stations, the busy extension is provided.
- For incoming calls to a principal with bridges, the principal's extension is provided.

- If the busy party is on the PBX, then the extension of the party is passed (if there is an internal error in the extension, a default value of [*****] is passed).

Call_id This switch-assigned call identifier is used to associate event reports to calls and to identify a call the adjunct wants to control. The call identifier is unique within the switch.

Cause Contains the reason the report is being sent. The following cause values are valid for this event: User Busy (CS0/17), No trunks available (CS3/20), No channel available (CS0/6), and Queue full (CS3/22).

Call Conferenced Event Report

The Call Conferenced Event Report is sent under the following circumstances:

- When an on-PBX station completes a conference by pressing the ‘‘conference’’ button on the voice terminal
- When an on-PBX station completes a conference after having activated the ‘‘supervisor assist’’ button on the voice set
- When the on-PBX analog set user flashes the switch hook with one active call and one call on conference and/or transfer hold
- When an application processor successfully completes a Third Party Merge request (conference option). The association over which the Third Party Merge request was made receives an acknowledgement; no Call Conferenced Event Report is sent over this association. All other associations controlling or monitoring the resulting call, including any other domain (station) control association(s) for the parties on the call, receive the Call Conferenced Event Report.
- When the ‘‘call park’’ feature is used in conjunction with the ‘‘conference’’ button on the voice set

Report Items

The following is a list of items provided with this report:

Calling Party number (controlling party number) The controlling/conferencing extension in the conference

Called Party Number (new party number) The newly conferenced-in extension

- If the newly conferenced-in party is an on-PBX extension, that extension is passed.
- If the party being conferenced in is off the PBX, then a default value of (#####) is passed.

> **NOTE:**
> There are scenarios in which the conference operation joins multiple parties to a call. In such situations, the destination extension is the extension for the last party to join the call.

Other Call_id The call identifier that ended as a result of the two calls merging

Resulting Call_id The call identifier retained by the switch after the two calls are merged

Party_id List (up to six numbers)	The party identifier is a number that corresponds to a specific extension number in the Extension List. The party list contains all party identifiers for all parties active on the call as a result of the conference.
Extension List (up to six entries)	The list of all extensions on the call. The extension consists of local PBX or group extensions. Group extensions are provided when the conference is to a group and the conference completes before the call is answered by one of the group members (TEG, PCOL, hunt group, or VDN extension). It may contain alerting extensions or group extensions as well as the default trunk values (if the call contains external parties). If the extension is on-PBX and there is an internal error in the extension, a default value of (*****) is passed. If the party is off-PBX, the default value of (#####) is passed.

Call Ended Event Report

The Call Ended Event Report applies to Event Notification associations only and is generated under the following circumstances:

- When the last party on a call drops
- When the switch cannot continue to send event reports for the call over an Event Notification association because the call has been merged (conferenced/transferred)

NOTE:
The Call Ended Event Report should not be confused with the Third Party Call Ended capability provided by the Call Control Capability group. The Call Ended Event Report and the Call Ended capability carry the same information. They differ in that the former allows the Event Notification association to continue and the latter terminates a Call Control association.

Also, the Call Ended Event Report is sent **only** over Event Notification associations; it is **never** sent over Call Control or Domain (Station) Control associations.

Report Items

The following is a list of items provided with this report:

Call_id The switch-assigned call identifier of the call that ended

Cause Contains the reason for the call ending:

- Normal clearing (CS0/16)
- Call with requested identity has been terminated (CS3/86)
- Call merged or Intercept SIT treatment — Number Changed (CS0/22)
- Answering Machine Detected (AMD) (CS3/24) |
- Non-ISDN endpoint dropped out of a connection (CS0/127) |

Call Initiated Event Report

The Call Initiated Event Report is sent by the switch to an adjunct that has a Domain Control association for a station under the following circumstances:

- When that station begins to receive dial tone
- When that station is forced off-hook because another ASAI association has requested a Third Party Make Call or a Third Party Auto Dial capability for the station. The switch does not send this event report to the requesting association if that association receives an acknowledgement (ACK) for the Third Party Auto-Dial request.

Report Items

The following is a list of items provided with this report:

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Call Offered to Domain Event Report

The Call Offered to Domain Event Report is generated when a call enters a domain (VDN or ACD Split) for which event reporting has been requested by the adjunct. This event only applies to event notification associations.

Call Offered to a VDN Domain:

The event is sent when a call enters a VDN domain that has notification active. From this point onward, call events for the call are reported.

If a call passes through several VDNs with notification active, then a Call Offered to Domain Event Report is generated for each such VDN over their respective monitoring association.

For G3V3 and later, a maximum of three notification associations can get events for a call. All associations receive the same events and event contents for calls entering the domain when the notifications are active.

Call Offered to an ACD Split Domain:

This event report is generated when a call enters an ACD split domain for which event reporting has been requested by the adjunct.

This report is sent even if the ACD split is in night service or has call forwarding active.

If a call passes through several ACD split domains with notification active, then a Call Offered to Domain Event Report is generated for each such ACD split domain over their respective monitoring associations.

Report Items

The following is a list of items provided with this report:

Calling Party Number/Billing Number (CPN/BN)

- For incoming calls over PRI facilities — ''calling number'' as in the ISDN SETUP message
- For incoming calls over non-PRI facilities, the calling party number is generally *not* provided. In this case, the Trunk Group number is provided instead.
- For calls originated at a bridged call appearance — the principal's extension
- For incoming DCS calls, if the DCS calling party information is available to the switch (if a station with a display gets it), this information is also made available to ASAI. Otherwise, the calling party information is provided as the default.

NOTE:
There is a special case of a switch-classified call being offered to a split. In this case, the Calling Party number contains the original digits (from a switch-classified Third Party Make Call Request) provided in the destination field.

- The Trunk Group number and Trunk Group Member number are only provided if the Calling Party number is unavailable.

Called Party number (DNIS)

- For incoming calls over PRI facilities, the Called Party Number as in the ISDN SETUP message.
- For incoming calls over PRI facilities to a VDN that does lookahead interflow on calls, if the lookahead interflow attempt fails, the called number provided is the principal extension of the dialed number.
- For incoming calls over non-PRI facilities, the Called Party Number is the principal extension [may be a group (TEG, PCOL, hunt group, VDN) extension[3]].

NOTE:
The special case of a switch-classified call being offered to a split, noted previously, contains the split extension as provided in the origination field of the Third Party Make Call Request.

User-Entered Information Contains up to 16 digits from the most recent Collect Digits vector step. For more information regarding vectors, see the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-204, and the *DEFINITY Communications System Call Vectoring/EAS Guide*, 555-230-520.

Lookahead Interflow Information If present, this information is passed unchanged to the adjunct as received by the destination switch. If information is not present, no information is passed.

Domain The extension of the monitored domain.

User to User Information If a call with UUI is delivered to a monitored domain (ACD split or VDN), the switch includes the UUI stored with the call in the Call Offered to Domain Event Report. This applies to UUI that originated in an ISDN PRI/BRI SETUP message; in

3. If the switch is administered to modify the DNIS digits, then the true DNIS is not passed.

an ISDN DISCONNECT message; or in an ASAI Route Select, 3rd Party Make Call, or 3rd Party Auto-Dial.

Flexible Billing Specifies that the billing rate can be changed for an incoming 900-type call. Present if the feature is allowed for the call and the Flexible Billing customer option is assigned to the switch.

Call Originated Event Report (G3V4)

The Call Originated Event Report is sent to notify the adjunct that the originating extension is attempting to establish a call. This indication is provided to ASAI applications monitoring the call through domain control only. The report provides the dialed digits. Link version 2 must be active. Two instances will not generate this report: TAC dialing over an ISDN trunk, and COR restrictions.

Report Items

The following is a list of the items provided with this report:

Call_id	[mandatory] This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.
Party_id	[mandatory] The switch-assigned identifier that uniquely identifies a party on a call; in this case, the party that originated the call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.
Calling Party Number	[mandatory] The number that originated the call; usually the extension that dialed the call. For calls originating from a Logical Agent, this is the logical agent number of the agent logged into the station making the call. For calls originating from a bridged call appearance, this is the number of the bridged appearance where the origination occurred.
Called Party Number	[mandatory] The number that the user dialed or requested by means of a Make Call or Auto Dial request.[4]
Party Extension	[optional] The originating device. Normally, this is the same as the calling number; however, in the case where a call is originated from a logical agent extension, it indicates the physical extension from which the call was made. For calls originating from a bridged call appearance, this is the number of the primary extension on the phone where the call originated.
UUI	[optional] This parameter is not included.

4. This number is represented before any digit manipulation, but does not include ARS, FAC, or TAC. The contents of this IE may change in subsequent Event Reports generated for a specific call. |

Cause [optional] This parameter is not included.

Call Redirected Event Report

The Call Redirected Event Report is sent to notify the adjunct that event reporting for a call will no longer be provided. This event report is sent under the circumstances detailed below.

For Monitored Calls

This event is sent when a monitored call enters a new domain that has Event Notification active. For example, if a call leaves one monitored domain and enters another, a Call Redirected Event Report is sent to the association that the call left. The Call Offered to Domain Event Report must have been received prior to the Call Redirected Event Report.

For G3V3 and later (Multiple Monitors feature), when an Active-Notification Call enters an Active-Notification Domain, the switch sends a Call Redirected Event Report over the notification association(s) that were active for the call and sends further Event Reports over the new notification association(s).

For Controlled Calls

This event report is sent over a Domain (Station) Control association when a call leaves the station, without the call having been dropped/disconnected. The Alerting Event Report must have been received prior to the call Redirected Event Report.

- When a call that had been alerting at the station leaves the station because:
  - One member of a coverage and/or answer group answers a call offered to a coverage group. In this case, all other members of the coverage and/or answer group that were alerting for the call receive a Call Redirected Event Report.
  - A call has gone to AUDIX coverage and the Coverage Response Interval (CRI) has elapsed (the principal call is redirected).
  - Principal answers the call while the coverage point is alerting and the coverage point is dropped from the call.
  - Stations are members of a TEG group with no associated TEG button (typically analog stations). |
  - The call was redirected using the ASAI Redirect Call capability. | However, if the Redirect Call request was sent over a domain | control association, then that domain association receives an Ack, | but does not receive the Call Redirected Event Report.
- When the domain-controlled station is an analog phone and an alerting call is now alerting elsewhere (gone to coverage or redirected by ASAI Redirect Call), either:
  - The pick-up feature is used to answer a call alerting an analog principal's station, or

— An analog phone call is sent to coverage due to ‘‘no answer’’ (the analog set’s call is redirected).

This event report *will not* be sent if the Domain (Station) Control is never alerted or if it retains a simulated bridged appearance until the call is dropped/disconnected. Examples of situations when this event is **not** sent are:

- Bridging
- Call forwarding
- Calls to a TEG (multifunction set with TEG button)
- Cover-All
- Coverage/Busy
- Incoming PCOL calls (multifunction sets)
- Pick-up for multifunction set principals

This event report never follows a Connected Event Report and is always preceded by an Alerting Event Report.

Report Items

The following is a list of the items provided with this report:

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Call Transferred Event Report

The Call Transferred Event Report is sent under the following circumstances:

- When an on-PBX station completes a transfer by pressing the ‘‘transfer’’ button on the voice terminal
- When the on-PBX analog set (phone) user on a monitored call goes on hook with one active call and one call on conference/transfer hold
- When the ‘‘call park’’ feature is used in conjunction with the ‘‘transfer’’ button on the voice set
- When an adjunct successfully completes a Third Party Merge request (transfer option). The association over which the Third Party Merge request was made only receives an acknowledgement; no Call Transferred Event Report is sent over this association. All other associations controlling or monitoring the resulting call (including the Domain Control associations) receive the Call Transferred Event Report.

Report Items

The following is a list of the items provided with this report:

Calling Party number (controlling party number)
The controlling/transferring extension in the transfer

Called Party Number (new party number)
The party that the call is being transferred to

- If the new transferred-to party is an extension on the local PBX, the extension is passed.

  If the party transferred-to is off the PBX, then a default value of (#####) is passed.

> **NOTE:**
> There are scenarios in which the conference operation joins multiple parties to a call. In these cases, the Called Party Number is the extension for the last party to join the call.

Other Call_id
The call identifier that ended as a result of the two calls merging

Resulting Call_id
The call identifier retained by the switch after the two calls are merged

Party_id List (up to six numbers)	The party identifier is a number corresponding to a specific entry number in the Extension List
Extension List (up to six entries)	The list of all transferred extensions on the call resulting after the transfer. It may contain alerting extensions or group extensions as well as default trunk values (#####) (if the call contains off-PBX parties). The extension consists of local extensions or group extensions. Group extensions are provided when the transfer is to a group and it takes place before the call is answered by one of the group members. If it is an on-PBX extension and there is an internal error in the extension, a default value of (*****) is passed.

Connected Event Report

The Connected Event Report is sent as follows:

- When a switch-classified call is delivered to an on-PBX party (after having been answered at the destination) and the on-PBX party answers the call (picked up handset or connected after zip tone)
- When a call is delivered to an on-PBX party and the on-PBX party has answered the call (picked up handset or cut-through after zip tone)
- When a call is redirected to an off-PBX destination, and the ISDN CONNect message is received from an ISDN-PRI facility
- Any time a station is connected to a call (for example, pick up on bridged call appearance, service observing, or busy verification)
- After a Trunk Seized Event Report for non-ISDN trunks

In general, this event report is not sent for split or vector announcements or attendant group 0.

Multiple Connected Event Reports

Multiple event reports may be sent for a specific call. For example, when a call is first picked up by coverage, the event is sent to the active associations for the coverage party, as well as to the active associations for all other extensions already on the call. If the call is then bridged onto by the principal, the Connected Event Report is then sent to the associations for the principal, as well as to the associations for all other extensions active on the call. For more information on bridging, see Chapter 12, ‘‘Feature Interactions.’’

Multiple event reports may also be sent for the same extension on a call. For example, when a call is first picked up by a member of a bridge, TEG, or PCOL, a Connected Event Report is generated. If that member goes on-hook and then off-hook again while another member of the particular group is connected on the call, a second Connected Event Report is sent for the same extension. This event report is not sent for split or vector announcements or attendant group 0.

Report Items

The following is a list of items provided with this report:

Calling Party number/Billing number (CPN/BN)

- For outgoing calls over PRI facilities — ‘‘calling number’’ as in the ISDN SETUP message
- For incoming calls over PRI facilities — ‘‘calling number’’ as in the ISDN SETUP message
- For incoming calls over non-PRI facilities the calling party number is generally *not* provided. In this case, the Trunk Group number is provided instead.
- For calls originated at a bridged call appearance — The principal’s extension

- For calls originated at a station — The station's extension
- For incoming DCS calls, if the DCS calling party information is available to the switch (if a station with a display gets it), this information is also made available to ASAI. Otherwise, the calling party information is provided as the default.

Trunk Group number and Trunk Group Member number

- The Trunk Group number and Trunk Group Member number are only provided if the Calling Party number is unavailable.

Called Party number (DNIS)

- For outgoing calls over PRI facilities, the called number is the one included in the SETUP message.
- For incoming calls over PRI facilities, the called number is the one provided in the SETUP message.
- For incoming calls over PRI facilities to a VDN that does lookahead interflow on calls, if the lookahead interflow attempt fails, the called number provided is the principal extension of the dialed number. If the interflow attempt is successful, the Called Party number is provided as the default.
- For incoming calls over non-PRI facilities, the called number is the principal extension (may be a group [TEG, hunt group, VDN] extension.[5])
- For incoming calls to PCOL, the called number is the default extension value (*****).
- For incoming calls to a TEG (principal) group, the TEG group extension is provided.
- For incoming calls to a principal with bridges, the principal's extension is provided.

Connected Party number

- For outgoing calls over PRI facilities—''connected number'' from the ISDN CONNect message
- For incoming calls — Locally connected extension (primary extension for TEGs, PCOLs, bridging)

5. If the switch is administered to modify the DNIS digits, then the modified DNIS is passed.

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Cause Contains the reason the report is being sent. The following cause values are valid for this event:

- Normal Answer (Answer Supervision from the Network, Internal Answer) (CSO/16)
- Timed Answer (Assumed Answer Based On Internal Timer) (CS3/17). Timed Answer is administered on the Trunk Group form.
- Voice Energy Answer (Voice Energy Detected By Classifier) (CS3/18)

Cut-Through Event Report

The Cut-Through Event Report is sent when an ISDN PROGress message has been received for a call using the ISDN-PRI facilities. The switch maps the PROGress message to a Cut-Through Event Report.

An ISDN-PRI network may send the switch a PROGress message for several reasons that are contained in the Progress Indicator (within the PROGress message). The switch forwards the PROGress indicator to the adjunct in the Cut-Through Event Report.

The switch may receive multiple PROGress messages for any given call; each maps into a Cut-Through Event Report for the call.

This event applies to all monitored or controlled calls except switch-classified.

This event is only sent to the destination party as follows: When a monitored or controlled call moves to an off-PBX destination and the PROGress message is received from the ISDN-PRI facility network.

Report Items

The following is a list of items provided with this report:

Call_id — This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id — The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Progress Indicator — Indicates interworking of other events occurring in the network.

Disconnect/Drop Event Report

A Disconnect/Drop Event Report is generated for a party that disconnects from a call. This event report is not generated over monitoring and control associations for the last disconnected party on a call; a Call Ended Event Report (monitoring association) or a Third Party Call Ended (control association) is generated instead.

A Disconnect/Drop Event Report is generated in the following cases:

- When a simulated bridged appearance is dropped (when one member drops)
- When an on-PBX party drops from a call
- When an off-PBX party drops and the ISDN-PRI receives a disconnect message
- When an off-PBX party drops and the non-ISDN-PRI trunk detects a drop

A Disconnect/Drop Event Report is *not* generated in the following cases:

- When a party drops as a result of a transfer operation
- When a split or vector announcement drops
- When an attendant drops off a call, if the call was received through attendant group 0
- When a switch-classified call is dropped during the call classification stage. (A Third Party Call Ended is generated instead.)
- When an association that invokes a Third Party Selective Drop or a Third Party Clear Call capability receives an acknowledgement and does not receive a Disconnect/Drop Event Report for the party or parties disconnected
- When a call is delivered to an agent and dequeued from multiple splits as part of vector processing

NOTE:
If a call has User to User Information (UUI) that came from an ISDN DISCONNECT message or in a Third Party Drop Request stored with it, and a party drops from the call and a Drop Event Report is sent, then the UUI is included in the report.

Report Items

The following is a list of items provided with this report:

Connected Party number (dropped number)	For an off-PBX party — The default trunk value of (#####) will be passed. If the party being dropped is local to the PBX, then the extension of the party being dropped is passed (primary extension for TEGs, PCOLs, bridging).

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

User to User Information If a call has User to User Information (UUI) that came from an ISDN DISCONNECT message or in a Third Party Drop Request stored with it, and a party drops from the call and a Drop Event Report is sent, then the UUI is included in the report. UUI is not sent if it did not come from either DISCONNECT or Third Party Drop.

Cause Contains the reason for the disconnection/drop. Any cause value passed from the network is included in an ASAI Drop Event Report. For ISDN endpoints, the cause value sent to the adjunct is the cause value received over the ISDN facility. In addition, CS0/127 is a normal drop cause sent for other than ISDN endpoints. For all cause values, see the *AT&T ISDN Primary Rate Interface Specification*, July 1989, 801-802-110.

Entered Digits Event Report

The Entered Digits Event Report is sent when a DTMF tone is detected because an inbound caller entered digits. The tone is disconnected when the far end answers or ‘‘#’’ is detected. Up to 24 digits can be collected. The digits reported include: 0-9, ‘‘*,’’ and ‘‘#.’’ The digit string includes the #, if present.

Report Items

The following is a list of items provided with this report:

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control.

Digits Entered Specifies the digits (in ASCII) collected as DTMF digits by the call prompter/tone detector.

Hold Event Report

The Hold Event Report is generated in the following case:

- When an on-PBX station places a call on hold (that is, hold or conference/transfer hold)

> **NOTE:**
> A call can be placed on hold either manually at the station or via a Third Party Selective Hold capability. When a call is manually placed on hold (that is, either by a hold, switch hook flash, or a conference/transfer hold), the event report is sent to all associations receiving event reports on that call.

> **NOTE:**
> An association requesting Third Party Selective Hold (that is, the association over which the capability was requested) receives an acknowledgement and does not receive a Hold Event Report; other associations monitoring the call receive the Hold Event Report.

Report Items

The following is a list of items provided with this report:

Connected Party number (number that placed the call on hold)	The extension that placed the call on hold
Call_id	This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.
Party_id	The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Login Event Report

This event report is generated when an agent logs into a Domain Control ACD split/skill. It is only provided with Link Version 2.

The agent may request login by using the feature access code/feature button or the agent may request login through ASAI and receive a positive ACK over the requesting association. The login event is sent over the domain control (ACD split) association.

Report Items

The following is a list of items provided with this report:

Agent Work Mode	The agent's work mode entered (AUX in most cases)
Agent Physical Extension	The physical extension of the agent logging in (EAS and ACD environments)
Agent Logical Extension	The logical extension of the agent logging in (EAS environment only)

Logout Event Report

This event report is generated when an agent is logged out of a Domain Control ACD split.

The adjunct may request agent logout by the Request Feature capability, or the agent may request logout by using the feature access code/feature button. The adjunct requesting logout by the Request Feature capability receives a positive ACK over the requesting association, independent of the logout event which is sent over the domain control (ACD split) association.

Report Items

The following is a list of items provided with this report:

Agent Physical Extension	The physical extension of the agent logging out (EAS and ACD environments)
Agent Logical Extension	The logical extension of the agent logging out (EAS environment only)

Queued Event Report

The Queued Event Report is sent as follows:

- When a switch-classified call is delivered to a hunt group or ACD split and the call queues
- When a call is delivered or redirected to a hunt group or ACD split and the call queues

It is possible to have multiple Queued Event Reports for a call. For example, the call vectoring feature may queue a call in up to three ACD splits at any one time. In addition, the event is sent if the call queues to the same split with a different priority. This event report is *not* sent when a call queues to an announcement, vector announcement, or Trunk Group. It is also not sent when a call queues, again, to the same ACD split at the same priority.

Report Items

The following is a list of items provided with this report:

Called Party number (DNIS) For incoming calls over PRI facilities, the destination number as in the SETUP message

For incoming calls over PRI facilities to a VDN that does lookahead interflow on calls, if the lookahead interflow attempt fails, the called number provided is the principal extension of the dialed number.

For incoming calls over non-PRI facilities, the called number is the principal extension [may be a group (TEG, PCOL, hunt group, VDN) extension.[6]]

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Calls in Queue This is the call position in the queue in the hunt group or ACD split. This number includes the current call and excludes all direct-agent calls in the queue.

Domain (ACD split) The extension of the ACD split to which the call queued

6. If the switch is administered to modify the DNIS digits, then the modified DNIS is passed.

Reconnected Event Report

This event report is generated under the following circumstances:

- When an on-PBX station connects to a call that has been previously placed on hold. Reconnecting to a held call can be done either manually at the station (by reselecting the active call appearance or switch hook flash) or via a Third Party Reconnect capability.

  NOTE:
  An association requesting Third Party Reconnect capability (the association over which the capability was requested) receives an acknowledgement and does not receive a Reconnected Event Report. Event reports are sent over the other associations monitoring and/or controlling the call.

Report Items

The following is a list of items provided with this report:

Connected Party number	The extension that has been reconnected to the call
Call_id	This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.
Party_id	The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Reorder/Denial Event Report

The Reorder/Denial Event Report is generated:

- When a call is trying to terminate to an on-PBX destination but the destination specified is inconsistent with the dial plan, has failed the ‘‘class of restriction’’ check, or inter-digit timeout has occurred
- When a call encounters a step in vector processing that causes the denial treatment to be applied to the originator
- When a direct-agent call is placed to a destination agent who is not a member of the specified split
- In a non-EAS environment, when a direct-agent call is placed to a destination agent who is not logged in
- In an EAS environment, when a direct-agent call is placed to a logical agent ID that doesn’t have a coverage path and is not logged in

Report Items

The following is a list of items provided with this report:

Called Party number — If the destination is inconsistent with the dial plan, the default extension value is sent (*****).

Call_id — This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Cause — Contains the reason for the reorder and/or denial

For this event report, any and all standard ISDN/PRI cause values are valid and are passed intact from the network to the adjunct. However, the following cause values are also explicitly sent from the switch:

Call rejected (CS0/21)
Invalid number (CS0/28)
Outgoing calls barred (CS0/52)
Incompatible destination (CS0/88)

ASAI-specific cause values for this report are: Agent not member of split (CS3/11) and Agent not logged in (CS3/15). Both cause values are for direct-agent calls.

Trunk Seized Event Report

The Trunk Seized Event Report is sent as follows:

- When a call is placed to an off-PBX destination and a non-PRI trunk is seized
- When a call is redirected to an off-PBX destination and a non-PRI trunk is seized

A Trunk Seized Event Report is *not* generated for outbound calls that use ISDN-PRI facilities, or for the destination of a switch-classified call.

Report Items

The following is a list of items provided with this report:

Called Party number Always the default trunk value (#####)

Call_id This switch-assigned call identifier is used to associate event reports to calls and identify a call the adjunct wants to control. The call identifier is unique within the switch.

Party_id The switch-assigned identifier that uniquely identifies a party on a call. The switch provides the identifier and the adjunct should retain it for future operations. The party identifier is unique within the call_id.

Use of Event Reports in Associations

The following table shows the type of association(s) over which various event reports can be sent.

Table 3-1. Use of Event Reports in Associations

Event Report	Domain (Station) Control	Call Control	Notification	Domain (Split) Control
Alerting	YES	YES	YES	NO
Answered	YES	YES	NO	NO
Busy/Unavailable	YES	YES	YES	NO
Call Conferenced	YES	YES	YES	NO
Call Ended	NO	NO	YES	NO
Call Initiated	YES	NO	NO	NO
Call Offered to Domain	NO	NO	YES	NO
Call Originated	YES	NO	NO	NO
Call Redirected	YES	NO	YES	NO
Call Transferred	YES	YES	YES	NO
Connected	YES	YES	YES	NO
Cut-Through	YES	YES	YES	NO
Disconnect/Drop	YES	YES	YES	NO
Entered Digits	NO	YES	YES	NO
Hold	YES	YES	YES	NO
Login	NO	NO	NO	YES
Logout	NO	NO	NO	YES
Queued	YES	YES	YES	NO
Reconnect	YES	YES	YES	NO
Reorder/Denial	YES	YES	YES	NO
Trunk Seized	YES	YES	YES	NO

Event Reporting for Merging Two Calls

This section covers the rules for sending events when two calls with different characteristics are merged for a transfer or a conference. For the purpose of these rules, it does not matter how the calls were merged (manually or via the Third Party Merge capability).

Call merging rules are as follows:

1. Merging two Call-Controlled calls
   - A Third Party Call Ended capability is sent for the call identifier that has been released as a result of the merge on the association that is controlling the call.
   - The Call Transferred or Call Conferenced Event Report is sent over the association controlling the resulting call identifier.
   - Subsequent event reports for the merged call are sent over the Call Control association controlling the resulting call identifier.
2. Merging two monitored calls (Event Notification is active)
   - A Call Ended Event Report is sent over the association monitoring the call for the call identifier that has been dropped.
   - The Call Transferred or Call Conferenced Event Reports are sent to the association monitoring the resulting call identifier.
   - Subsequent event reports are sent over the Event Notification association monitoring the resulting call identifier.
3. Merging one Call-Controlled call with a nonmonitored or controlled (Event Notification or Call Control are not active) call
   - The Call Transferred or Call Conferenced Event Reports are sent over the Third Party Call Control association. This association continues to report subsequent events for the remaining call.
4. Merging one monitored call (Event Notification active) with a nonmonitored or controlled (Event Notification and Third Party Call Control are not active) call
   - The Call Transferred or Call Conferenced Event Reports are sent over the Event Notification association.
   - Subsequent event reports are sent over the Event Notification association of the monitored call. In the case where the resulting call is the nonmonitored call, the Call Conferenced or Call Transferred Event Report is the very first event the adjunct receives with a new call identifier. If the resulting call identifier is from the Call-Controlled call, the adjunct receives the Call Conferenced or the Call Transferred Event Report(s).
5. Merging one Call Control association and one monitoring association
   - Both associations are retained and continue to provide subsequent event reports for the resultant call.

- The Event Notification association could receive a Call Ended Event report for the call that has been released as a result of the merge. This only occurs if the call being released is a monitored call.
- Both associations receive a Call Conferenced or Call Transferred Event Report.

6. Merging two calls with Domain (Station) Control extensions
   - Subsequent event reports for the resulting call are provided for the Domain (Station) Control associations for extensions remaining on the call.
   - A Call Conferenced or Call Transferred Event report is provided to add Domain Control associations.
7. Merging one Domain (Station) Control and one Third Party Call Control association
   - All subsequent event reports for the resultant call are provided to both associations.
8. Merging two calls, one of which is not monitored or controlled, and the other containing only Domain (Station) Control extension(s)
   - Subsequent event reports are provided for the Domain (Station) Control association for the extensions remaining on the resultant call.
9. Merging two calls with a combination of a Domain (Station) Control association and an Event Notification association
   - The Event Notification association could receive a Call Ended Event report for the call that has been released as a result of the merge. This only occurs if the call being released is a monitored call.
   - A Call Conferenced or Call Transferred Event report is provided to the Event Notification association and the Domain (station) Control association.
   - Subsequent event reports for the call are provided over the Event Notification association.
10. Merging two calls with a combination of Call Control, Event Notification, and Domain (Station) Control associations (but only one call control and only one Event Notification association between the two calls). Items 5, 7, and 9 apply to this case.

For G3V3 and later, the Multiple Monitors feature impacts transfer and conference situations as follows:

- If the resultant call (after the conference or transfer) has ANY notification associations active, the switch sends further Event Reports over the associations that were active for the resultant call. The switch also sends the Call Ended Event Report over the notification associations for the other call.

  Scenario 1: Call-1 has notification associations A1, A2. Call-2 has notification association A3. Call-2 is the resultant call.

  After the merge (transfer, in this case) events are sent as follows:

  A1: Call Ended
  A2: Call Ended
  A3: Call Transferred, further-events

- If the resultant call (after the conference or transfer) has NO notification associations active, the switch sends further Event Reports for the notification associations that were active for the other call. The Call Ended event is not sent over any notification associations.

  Scenario 2: Call-1 has notification associations A1, A2. Call-2 has NO notification associations. Call-2 is the resultant call.

  After the merge (transfer, in this case), events are sent as follows:

  A1: Call Transferred, further-events
  A2: Call Transferred, further-events

Table 3-2 provides a summary of the call merging rules described above.

Table 3-2. Call Merge Summary

Call Characteristics	Resultant Association(s)	Conf/Xfer Event Sent to:	Call Ended Event Sent to:
1. Call-Ctl & Call-Ctl*	Ctl Assoc	Resultant Assoc	Other Ctl Assoc
2. Monitored & Monitored	Monitored	Resultant Assoc	Other Monitored Assoc
3. Call-Ctl & non-Ctl & non-Monitored	Call-Ctl	Call-Ctl Assoc	-
4. Monitored & non-Ctl/non-Monitored	Monitored	Monitored	Monitored
5. Call-Ctl & Monitored	Both	Both	Monitored
6. Dom-Ctl & Dom-Ctl**	Both	Both	-
7. Dom-Ctl & Call-Ctl**	Both	Both	-
8. non-Ctl non-Monitored Dom-Ctl	Dom-Ctl	Dom-Ctl	-
9. Dom-Ctl & Monitored	Both	Both	Monitored
10. Monitored/Ctl/Dom-Ctl	All	All	-

* Allowed via Third Party Merge on Call Control

** Allowed on a Third Party Merge of a Domain (Station) Control Association

Rules for Merging Two Calls with UUI Information

1. In transfer and conference scenarios in which UUI is associated with both calls, the UUI associated with the resulting call is kept for the new call.
2. In transfer and conference scenarios in which UUI is associated with either one of the two calls, the UUI is associated with the resulting call.
3. If a call with UUI is held to initiate a transfer or conference, and the second call is placed without UUI, then the following occurs: If the event is sent before the two calls are merged by the completion of the conference or transfer, an event report or ISDN message sent about the second call will not include UUI.

U-ABORT

This capability is used by both the adjunct and the switch to inform the other that an association is abnormally ending. It is shown here but may be used in all capability groups at any time during an association.

Information Flow

The U-Abort ends the association and no response is possible.

U-Abort Parameters

The switch may issue one of the following reasons as the cause for ending the association:

- Error in vector/host not responding (CS0/102)
  Either the vector is incorrecly programmed or the host has not responded to the route request in a timely manner.
- Protocol error (CS0/111)
  The Q.932 protocol has been violated.
- Call merge attempt failed (CS0/98)
  An attempt was made to merge two calls that were not controlled.
- Error in association (CS3/87)
  The switch software audits found something wrong and the switch is taking steps to correct the association.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *AT&T DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221.

ASAI and Call Control

4

This chapter describes the Call Control Capability Group.[1] The capabilities in this group are categorized by their functions: initiating, controlling, and terminating. The adjunct uses these capabilities to set up (place), take control of, monitor, or end a call on the switch.

Third Party Make Call

This initiating capability enables the adjunct to set up a call on behalf of an on-PBX station. The four types of controlled calls supported by this capability are as follows:

Switch-classified

User-classified (with or without priority)

Direct-agent

Supervisor-assist

Third Party Take Control

This initiating capability enables the adjunct to take control of a call that is already in progress to perform additional call control functions when needed.

Third Party Selective Hold

This controlling capability enables the adjunct to place the controlled call on hold at the specified station.

1. Capability groups are analogous to the Application Service Elements (ASEs) described in the *AT&T Adjunct/Switch Application Interface (ASAI) Specification*.

Third Party Reconnect	This controlling capability enables the adjunct to direct the switch to reconnect a specified station to the controlled call.
Third Party Merge	This controlling capability enables the adjunct to direct the switch to merge two controlled calls at a specified station into one call on behalf of that party.
Third Party Selective Drop	This controlling capability enables the adjunct to direct the switch to drop a party from the controlled call.
Third Party Relinquish Control	This terminating capability enables the adjunct to relinquish control of a specific call and direct the switch to stop sending event reports for this call.
Third Party Clear Call	This terminating capability enables the adjunct to direct the switch to disconnect all parties on a controlled call.
Third Party Send DTMF Signals	This capability enables the adjunct to transmit a sequence of DTMF tones on behalf of a party on the call.
Third Party Call Ended	This terminating capability is used by the switch when all parties on a controlled call are disconnected.
Redirect Call	This capability enables the adjunct to direct the switch to redirect an alerting call away from the alerting station extension and route it to another number.
Event Reports	See Chapter 3, ''Common Capabilities.''
U-Abort (User Abort)	See Chapter 3, ''Common Capabilities.''

Call Control capabilities are accepted only when the call is in the proper call state with respect to the specified party. The following table specifies which call control requests are accepted and which are denied, depending on the state of the call or the affected party. In addition, the switch restricts Call Control requests based on the call originator's and destination's Class of Restriction (COR).

Table 4-1. Call Control Acceptance in Various Call/Station States

Call Control Request	Active St	Held St	Alerting St	Idle St	Dialing St	Bridged St
Third Party Make Call	no	yes[1]	yes[1]	yes	yes[2]	yes[1]
Third Party Take Control	yes	yes	yes	3	yes	yes
Third Party Relinquish Control	yes	yes	yes	3	3	yes
Third Party Clear Call	yes	yes	yes	3	yes	yes
Third Party Selective Drop	yes	no	no	3	yes	no
Third Party Reconnect	yes	yes	no	3	no	yes
Third Party Merge[4]	yes	yes	no	3	no	no
Redirect Call	no	no	yes	no	no	no

[1] If the state of the other calls at the station is as specified and there is an idle call appearance

[2] Call appearance previously selected for call origination

[3] Call_id does not exist

[4] Request should be made over the held call association; if made over the active call association, it is denied.

Third Party Make Call

The adjunct uses this initiating capability to establish a call between two parties.

Information Flow

An acknowledgement (ACK) is optional for the Third Party Make Call capability. If requested, an ACK is sent to the adjunct when the switch successfully originates the call (a call identifier has been assigned). If conditions prohibit a call from being established, a Negative Acknowledgement (NAK) is returned.

Third Party Make Call Parameters

orig_addr — The calling station's address. For non-switch-classified calls, the request is denied unless the calling party is off-hook, idle, dialing, listening to dial tone, goes off-hook within five seconds of the request, or is forced off-hook by the switch (through a speakerphone). See ''Denial (NAK) Causes'' that follows.

dest_addr — The called party's address.

dest_route_select — [optional] The trunk access code or feature access code (for example, Automatic Route Selection [ARS], or Automatic Alternate Routing [AAR]) of a called endpoint that is not directly connected. Alternately, this information may be included in the **dest_addr**.

return_ack — [optional] If this parameter is present, it indicates that the switch should return an acknowledgement to the request. If this parameter is not present, then the switch does not return an acknowledgement.

call_options — Call options are listed below and described in detail in the Third Party Make Call overview later in this section.

- alert_order
- ans_mach
- anw_treat
- direct_agent_call
- max_ring_cycle
- priority_calling
- service_circuit
- supervisor_assist
- uui_info

ACK (Positive Acknowledgement) Parameters

If the **return_ack** is present in the adjunct's request, the switch returns an ACK containing the party_id, call_id, and connected_number_id of the originator on the call. If the **return_ack** is not present, no ACK is returned.

Denial (NAK) Causes

The following causes may be issued for non-switch-classified calls. For NAKs for switch-classified calls, see ‘‘Third Party Make Call and Supported Call Types’’ that follows.

- Invalid association (CS0/81)
  The association does not exist.
- Requested facility not subscribed/provisioned (CS0/50)
  The user has not subscribed for the requested capability.
- Out of Service (CS3/27)
  The originator of the call is out of service.
- Switching Equipment Congestion (CS0/42)
  The PBX equipment is experiencing congestion due to traffic overload.
- Call Rejected (CS0/21)
  The COR check for completing the call failed.
- Mandatory Information Element missing (CS0/96)
  A required parameter is missing in the request.
- Resources not available (CS3/40)
  - The request cannot be executed because a call classifier is not available or no trunks are available in the requested trunk group.
  - The maximum number of calls being controlled has been exceeded.
- Incompatible Options (CS3/80)
  The designated call options are not compatible with the Third Party Make Call request.
- User busy (CS0/17)
  The user is busy on another call and cannot originate this call.
- No user responding (CS0/18)
  The originating address does not pick up the call and cannot be forced off-hook.
- Invalid number/domain (CS0/28)
  An invalid address or extension number is present in the request.
- Agent not member of split (CS3/11)
  The agent is not a member of the split in this Direct Agent call.
- Agent not logged in (CS3/15)
  The agent is not currently logged into a split for this Direct Agent call.
- Bearer capability not presently available (CS0/58)
  There are incompatible bearer services for the originating or destination addresses. For example, the originator is administered as a data hotline station or the destination is a data extension.
- Answering Machine Detected (AMD) (CS3/24)
  The call was answered by an answering machine.

- Outgoing calls barred (CS0/52)
  The requested Third Party Make Call is being attempted over a trunk that has been restricted from use by the originator.
- Invalid Information Element contents (CS0/100)
  The call_id specified by the Third Party Take Control capability is beyond the maximum allowable call_id value. For example, you could get this cause value if the switch received a User to User Information (UUI) Information Element longer than 32 bytes in an ASAI message or one that did not include the header.
- Message not compatible with call state (CS0/98)
  The message is not compatible with the CRV state.
- Service or option not subscribed/provisioned (CS0/50)
  If an adjunct requests a Third Party Make Call (switch-classified call) with Answering Machine Detection and AMD is not enabled, the switch denies the request.

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The switch allows only one adjunct to control a call. Additional requests to control a call by other adjuncts are denied by the switch.

When a multifunction station user is receiving a dial tone on a call appearance and the adjunct invokes a Third Party Make Call for that station, the switch selects the same call appearance and call_id to establish the call.

The sum of the dest_addr and dest_route_select parameters can equal 32 digits.

Third Party Make Call and Supported Call Types

The Third Party Make Call capability is used by the adjunct to request the switch to set up a call between two parties (calling endpoint and the called endpoint).

Four types of controlled calls are supported for the Third Party Make Call capability:

1. Switch-classified call
2. User-classified call
3. Direct-agent call
4. Supervisor-assist call

1. Switch-Classified Call

A switch-classified call is the only call type used primarily by OCM applications for Predictive Dialing when the destination is alerted first (using the *alert_order parameter*). The administrator determines which calls are given to the agent based on voice energy or the returned SIT or AMD tones from the call. All other calls are dropped by the switch and the classification reported to the adjunct (via event reporting and/or call ended capabilities).

See Table 4-2 for SIT and AMD tone definitions.

This is the only type of call placed with the alerted order and the service circuit (classifier) options specified.

The originator of a switch-classified call must be a valid on-PBX extension number associated with a split, hunt group, announcement, or VDN. The originator cannot be a station extension. (However, in an EAS environment, a VDN extension must be the originator for a predictive dialing application. See the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520, for more information.)

The destination can be either a valid station extension (except Distributed Communications System [DCS] or Uniform Dial Plan [UDP] extension), or an off-PBX number. ISDN-PRI trunk types are the preferred facilities for these outbound calls. It is recommended that trunks with answer supervision be used for switch-classified outbound calls. Answer supervision from the network over the trunk facility provides an accurate and timely indication when the far end answers. Otherwise the switch relies on the call classifier to do the call classification.

Call Parameters for Switch-Classified Calls

This section contains the ASAI interface call parameters for switch-classified calls.

orig_addr	[mandatory] The originator must be a valid local extension number associated with a split, hunt group, or announcement, or a VDN in an EAS environment.
dest_addr	[mandatory] The destination can be a valid switch extension or an off-PBX destination. It may include TAC/ARS/AAR information.
dest_route_select	[optional] This parameter contains TAC/ARS/AAR information for off-PBX destinations if the information is not included in the dest_addr.
priority_calling	[optional] If present, a priority call is placed if the destination is an on-PBX extension. If the priority flag is specified for an off-PBX destination, the call is denied.
max_rings	[optional] This parameter designates the number of rings that are allowed before classifying the call as no answer. The minimum number is two, the maximum number is 15, and the default is 10 (if not present).
direct_agent_call	This parameter must not be present.
supervisor_assist	This parameter must not be present.
alert_order	[mandatory] This parameter must be set to indicate ‘‘destination first.’’
ans_mach	This parameter indicates if answering machine detection is desired or not. Values are ‘‘On’’ or ‘‘Off.’’ If ‘‘On’’ is selected, the anw_treat parameter specifies the treatment for an answering machine call.
anw_treat	This parameter is mandatory if ans_mach is ‘‘On.’’ It specifies the call treatment when an answering machine is detected. The values are ‘‘drop,’’ ‘‘answer,’’ or ‘‘blank.’’ If blank, the value defaults to the administered value.

service_circuit [optional] This parameter must be set to ‘‘classifier’’ if the destination is off-PBX and the trunk access code is contained in the dest_route_select.

This parameter is optional if the destination is on-PBX or the trunk access code is included in the dest_addr parameter. If the trunk access code is part of the destination address and no service circuit parameter is included, then answer may not be detected over non-ISDN trunks. This may lead to the call being hung for up to ten minutes if the call is not answered.

return_ack [optional] If this parameter is present, it indicates that the switch should return an acknowledgement to the request. If this parameter is not present, then the switch does not return an acknowledgement.

uui_info [optional] UUI information received in a successful Third Party Make Call on the ASAI link is stored by the switch with the call for the life of the call or until overwritten due to a later UUI IE associated with the call.

UUI from a Third Party Make Call will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, and in a Route Request, if one is made.

Call Classification

The switch uses the Call Classification process, along with a variety of internal and external events, to determine a switch-classified call’s outcome. Whenever the called endpoint is external, a call classifier is used.

The classifier is inserted in the connection as soon as all the digits have been outpulsed.

A call is classified as either ‘‘answered’’ or ‘‘dropped’’ (Third Party Call Ended).

Ringing is reported to the adjunct as an Alerting Event Report, but in and of itself is not a final classification. ‘‘Non-classifiable energy’’ is always treated as an answer classification and reported to the adjunct as such. The modem’s answer back tone results in a Third Party Call Ended. The SIT detection is reported to the adjunct as an Answered Event Report or Third Party Call Ended depending on the customer’s administration preference. Answering Machine Detection (AMD) is reported as answered or call ended, depending on administration or call options. See *DEFINITY Communications System Generic 3 V4 Implementation*, 555-230-655 for more information regarding administration of SIT reporting to the adjunct.

The conditions detected are as follows:

Table 4-2. Detected SITs

SIT Name	Cause Value Reported
SIT Reorder	CS0/42 Switching Equipment Congestion
SIT No Circuit	CS0/34 No Circuit or Channel Available
SIT Intercept	CS0/22 Number Changed
SIT Vacant Code	CS0/01 Unassigned Number
SIT Ineffective Other	CS0/28 Invalid Number/Domain
SIT Unknown	CS0/31 Normal Unspecified
AMD Treatment	CS3/24 Answering Machine Detected
Talk Duration (seconds)	---
Pause Duration (seconds)	---

Answer Classification

The Answer Classification applies only to the destination (either internal or external). For off-PBX destinations, the answer event at the far end can be determined either by the call classifier or the network answer supervision.

A call is considered answered if one of the following occurs (whichever is first):

- **Network Answer Supervision —** The network provides answer notification to the switch (for trunks supporting answer supervision). For outbound calls over ISDN-PRI facilities, the CONNect message is interpreted as ‘‘far end answered the call.’’ If Answering Machine Detection (AMD) has been requested for the call, the Network Answer Supervision (non-ISDN-facilities) and the CONNECT message (ISDN facilities) are ignored. For these cases, the call classifier determines whether a call has been answered, depending on the AMD treatment requested for the call.
- **Call Classifier —** The call classifier detects an answer when:
  - Voice energy is detected by the call classifier.
  - A ‘‘SIT-detected’’ message is received from the call classifier reporting a specific Special Information Tone (SIT) which must be treated as an answered call (administered option).
  - A ‘‘non-classifiable energy detected’’ message is received from the call classifier.

— The ‘‘timeout’’ message is received from the software and told by the call classifier reporting that there was no tone detected after a timeout interval. This information appears on the System Parameters-Features form as the ‘‘off-premises tone detect timeout.’’

— Answering Machine Detection (AMD) is enabled for a call and an Answering Machine is detected.

- **Local detection —** A call placed to an on-PBX endpoint is considered answered when the called party answers the call.

If the call is answered in one of the ways described above, the Answered Event Report is sent to the adjunct. When the call results in a SIT which is to be treated as answer, a cause value with the particular SIT is included in the Answered Event Report. See Chapter 3, ‘‘Common Capabilities’’ for more information on the Answered Event Report’s contents.

If an answering machine is detected, the call is treated according to administration or to treatment requested in the message. Treatment in the message overrides the administered treatment.

Drop Classification

The classifier is immediately disconnected following detection of one of the conditions described below.

No Answer Condition

For calls placed to off-PBX destinations, the ‘‘no answer’’ condition is the result of the call classifier detecting ringing energy for the entire duration specified (the maximum number of rings times six seconds). The number of rings for each call may be selected from a range (two to 15) provided as an option in the Third Party Make Call capability. Values outside this range will cause the call to be denied (cause CS3/79 — Service or Option Not Implemented). The default is 60 seconds (10 rings).

For calls placed over ISDN-PRI facilities, if the ISDN ALERTing message is received from the network, the classifier is expected to detect the ‘‘in band’’ ringing and provide the ‘‘no answer’’ classification independent of receiving an ISDN ALERTing message.

For calls placed to on-PBX destinations, the ‘‘no answer’’ classification is done by counting the number of seconds of ringing.

The adjunct is notified of the no answer outcome in a Third Party Call Ended with cause CS3/19 — No Answer.

Busy Condition

For calls placed to off-PBX destinations, the busy results from the call classifier detecting either precise or imprecise busy tones or the ‘‘DISConnect/cause=user busy’’ (CS0/17 — User Busy) message returned by an ISDN-PRI facility.

A Third Party Call Ended is sent in these cases to the adjunct with cause value CS0/17 — User Busy.

For calls placed to on-PBX destinations, the busy condition is the result of the ‘‘busy’’ internal feedback message. Again, a Third Party Call Ended is sent to the adjunct with cause value CS0/17 — User Busy.

Reorder Condition

For calls placed to off-PBX destinations, the reorder condition is from the call classifier detecting either precise or imprecise network reorder tones, or receipt of an ISDN ‘‘progress’’ or ‘‘disconnect’’ message with cause CS3/42 — Denial/Reorder. A Third Party Call Ended is sent to the adjunct (cause CS3/42 — Denial/Reorder).

For calls placed to on-PBX destinations, the reorder classification is the result of the ‘‘reorder’’ internal treatment. A Third Party Call Ended is sent with the following cause: CS0/34 — No Circuit or Channel Available.

Other Conditions

When a switch-classified call is made to a destination that gives modem tone, the |
call is dropped (ended) with cause CS0/58 — Requested Bearer Capability not |
Available.

Switch-Terminated Call Conditions Prior to Classification

Before a call is classified, some conditions will cause the switch to terminate the call attempt:

- No trunks available (none administered or none idle)
(cause CS3/20 — Trunks not available)
- Attempting to use a Trunk Access Code (TAC) to access a PRI trunk (only Automatic Alternate Routing/Automatic Route Selection [AAR/ARS] feature access codes may be used to place a switch-classified call over a PRI trunk) (cause CS0/21 — Call Rejected)
- No classifiers available
(cause CS3/21 — Classifiers not available)
- Resources not available (for example, time slots or association records) — NAK/Return Error component (cause CS3/40 — Resources not available)
- Receipt of a RELease COMplete message from the adjunct

Switch Operation for Switch-Classified Call Setup

This section details the switch actions for each stage of call setup.

After the initial alert event is reported, the switch waits for the call to be classified as answered or dropped. The Alerting Event Report is sent to the adjunct only for the first ringing occurrence.

If network answer supervision is received before the classifier detects answer, the classifier is disconnected. If the classifier detects the answer, any subsequent network answer supervision indications are ignored.[2]

After the answered event is reported (call answered at the called party), the switch terminates the call at the originator specified in the Third Party Make Call request. Valid originators are ACD split extensions, hunt group extensions, announcement extensions, or VDNs.

When the switch determines that the originator (ACD split) can accept the call, the call is transferred to the ACD split (the sequencing is done internal to the switch). The event report sent to the adjunct depends on whether the call has been delivered to an available agent/extension or has been queued. If the call drops prior to getting to the originator, the Third Party Call Ended Report is sent to the adjunct.

If the call has been queued, the Queued Event Report is sent to the adjunct. When the agent becomes available, the call is de-queued and an Alerting Event Report followed by a Connected Event is sent containing the extension receiving the call.

The Alerting and Call Connected Event Reports are sent to the adjunct when the call is being delivered at a physical endpoint (agent/extension). The Connected Party number item and its extension are included in these event reports. For announcements, the connected party extension is the same as that specified for the calling endpoint in the Third Party Make Call request. For hunt groups and splits, this is the member/agent extension receiving the call.

The Call Ended Event Report is sent to the adjunct when the call is dropped because of a busy, reorder, denial, answer-back tone, or no-answer result. A separate Dropped Event Report is not sent for these outcomes.

If a SIT was detected, the call is either dropped (call ended message is sent with cause=SIT value), or it is considered answered. See Table 4-2 for definitions of the SITs detected.

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2. See Answering Machine Detection operation for G3V3 and later switch releases in this section.

For G3V3 and later, an ASAI adjunct may request Answering Machine Detection (AMD) for a switch-classified call. If AMD is requested for a call, the receipt of Network Answer Supervision or the ISDN CONNECT message is no longer sufficient to classify the outcome of a call. Further classification by the call classifier is required.

If an answering machine is detected, a cause value is sent in either the Answered or the Call Ended message to indicate such, and the call is either dropped or connected. If the treatment selected is connect, an Event Report is sent to the adjunct. The treatment may be specified either system-wide or on a per-call basis.

NOTE:
When AMD is requested on a switch-classified call, the reporting of the call outcome is delayed because detection of voice energy is no longer sufficient to classify a call, and further classification is required to distinguish between a live answer and a machine answer. However, the maximum number of call classifiers required for switch-classified calls need not be increased, since the increase in average holding time for call classifiers is minimal (up to 2 seconds). AMD uses the following Call Classifiers: TN744B (or later), and TN2182.

For switch-classified calls, the COR of the ASAI link is used to determine whether a call can be made to the destination indicated in the request.

Switch-Classified Call Originator

When the destination answers, the originator receives ringing. If the originator intraflows with priority, the new destination receives priority ringing. The originator receives the zip tone if it is in auto-answer mode.

When the originator is a logged-in ACD agent, ACD call delivery rules apply. When the originator is a station user (the call was delivered through a hunt group, not an ACD split), normal alerting and call delivery take place.

If the originator has a display set, the display shows the destination’s extension (if the destination is internal), or the name of the trunk group (if the destination is external).

With the OCM/EAS feature enabled, a VDN extension can be an originator for a switch-classified call. See the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520, for more information.

It is recommended that agents receiving switch-classified calls work in auto-answer mode with headsets (administered with ‘‘auto-answer on’’).

Switch-Classified Call Destination

If the destination is on-PBX, the user receiving the call will receive a priority ring if it has been requested by the Third Party Make Call Request.

If the destination has a display, the originating group (for example, the hunt group) is displayed as the calling party. This call cannot be picked up by a pickup group user.

Negative Acknowledgement (NAK) of a Switch-Classified Call

A Third Party Make Call (switch-classified) request is denied if:

- The ‘‘type’’ field in the Domain IE (which codes the dest_route_select) does not specify ‘‘trunk’’ (cause=CS0/100).
- The switch cannot obtain a time slot or other internal resource (cause=CS3/40).
- The alert order is not specified as ‘‘alert destination first’’ (cause=CS0/100).
- The service circuit requested is not a call classifier (cause=CS0/100).
- The calling number is neither a split nor an announcement extension (cause=CS0/28).
- An adjunct requests a Third Party Make Call with Answering Machine Detection and AMD is not enabled (cause=CS0/50).
- The request specifies that the call is to be direct-agent or supervisor-assist as well as switch-classified (cause=CS3/80)
- A service circuit is not requested for external calls (cause=CS0/96)

A Third Party Make Call (switch-classified) is dropped if:

- The COR check for delivering the call has failed — Call Ended Event Report (cause=CS0/21).
- An answered call at the destination cannot be delivered because of a full queue condition. This condition is logged in the error log (CS3/22).

Once a switch-classified call is set up and connected to a station user, if the station is not locked, the station user may control the call (for example, place on hold, reconnect, transfer, conference, drop). The adjunct may also control this call using Third Party Call Control capabilities. The association used for the Third Party Make Call request by the adjunct is used for subsequent event reporting and/or call control.

2. User-Classified Call

This type of Third Party Make Call is used by many types of applications such as Office Automation, Messaging, and OCM applications for Preview Dialing. It is defined as a call that is originated from a station with all Direct-Agent or Supervisor-Assist call options turned off. The call is set up by the adjunct on behalf of a station extension (calling party) to an on- or off-PBX called endpoint.

The adjunct must provide the calling party origination and the called party destination addresses. Valid originators for this type of call include all station extensions. Valid destinations are on-PBX extensions (including VDNs) and off-PBX numbers. All trunk types (including ISDN-PRI) are supported as facilities for reaching called endpoints for outbound User-Classified calls. Call progress feedback messages are reported as events across the BRI ASAI interface. Answer supervision and call classifiers are not used for this type of call.

Parameters for User-Classified Calls

This list contains the ASAI interface call parameters for User-Classified calls.

orig_addr	[mandatory] Must be a valid station extension.
dest_addr	[mandatory] Must be a valid extension number or off-PBX number. An on-PBX extension may be a station extension, VDN, split, hunt group, or announcement extension. The dest_addr may include TAC/ARS/AAR information for off-PBX numbers.
dest_route_select	[optional] If present, it contains the TAC/ARS/AAR information for off-PBX numbers, if they were not present in the dest_addr.
priority_calling	[optional] If present, a priority call is placed if the destination is a local extension. If the priority flag is specified for an off-PBX destination, the call is denied. The default is nonpriority.
max_rings	N/A (ignored)
direct_agent_call	Must not be present
supervisor_assist	Must not be present
alert_order	[optional] If present must be NO (alert originator first) It is recommended that this item not be present.
service circuit	Must not be present
return_ack	[optional] If this parameter is present, it indicates that the switch should return an acknowledgement to the request. If this parameter is not present, then the switch does not return an acknowledgement.
uui_info	[optional] UUI received in a successful Third Party Make Call on the ASAI link is stored by the switch with the call for the life of the call or until overwritten due to a later UUI IE associated with the call.

UUI from a Third Party Make Call will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, and in a Route Request, if one is made.

Call Classification for User-Classified Calls

All call-progress audible tones are provided to the originating user at the calling endpoint (except that user does not hear dial tone or touch tones). Call progress feedback is sent to the adjunct in event reports. For OCM preview dialing applications, final call classification is done by the station user staffing the originating set (who hears call progress tones and manually records the result).

Switch Operation for User-Classified Call Setup

Before trying to set up the call, the switch validates the field/options consistency. It then attempts to originate the call for the originating user on behalf of the extension specified in the originating address (orig_addr) call parameter.

Once the call is successfully originated, the switch does not drop it regardless of outcome. The only exception is the denial outcome which results in the intercept tone being played for 30 seconds, after which the call is disconnected. The normal audible feedback is provided to the originating station user. It is then up to the originating station user or application to drop such calls (either by going on-hook or via a Call Control request). For example, if the adjunct places a call to a busy destination, the originator will be busy until he or she normally drops or until the adjunct sends a Call Control command to drop the call.

Once set up, either the station user or adjunct may control this call. The adjunct may then use the association used for the Third Party Make Call request to control the call.

User-Classified Call Originator

For the originator to place the call, the originator’s set (display or voice) must have an available call appearance for origination and must not be in the talking (active) state on any call appearances. The originator is allowed to have a call(s) on hold or alerting at the set.

The originator may go off-hook first, and then issue the Third Party Make Call request. The switch originates the call on the call appearance with dial tone.

If the originator is off-hook busy, the call cannot be placed and the request is denied (NAK/Return Error component, cause=CS0/17 — User busy). If the originator is unable to originate for other reasons, the switch denies the request. See ‘‘Negative Acknowledgement of a User-Classified Call’’ later in this chapter for more detail.

After dialing is completed, the calling endpoint hears call progress tones (but not dial tone or touch tones). If the call was placed to a VDN extension, the calling endpoint hears whatever has been programmed for the vector associated with that VDN. If the calling endpoint has a display set, the display shows the called endpoint's extension and name (if the called endpoint is on-PBX), or the name of the Trunk Group (if the called endpoint is off-PBX).

Originators may be ACD agents in various work modes.

User-Classified Call Destination

If the destination is an on-PBX station user, the user receiving such a call receives alerting (according to whether the call was priority or not). Normal display interactions apply for destinations with displays.

When the destination is a logged-in ACD agent or an ACD split, ACD call delivery rules and display features apply. A User-Classified call whose destination is an ACD agent's extension is delivered to that ACD agent like a personal call.

When the destination is a VDN extension, vector processing rules apply.

Announcement destinations are treated like on-PBX station users.

The station user receiving the call is alerted according to the call type (ACD, priority, or normal). Call delivery depends on the call type, station type, station administered options (for example, manual/auto answer or call waiting), and station's talk state.

For example, for an ACD call, if the user is off-hook idle and in auto-answer mode, the call is cut-through immediately. If the user is off-hook busy and has a multifunction set, the call alerts a free call appearance. If the user is off-hook busy and has an analog set and the user has ‘‘call waiting’’ or this is a priority call, the analog station user is given the ‘‘call waiting tone.’’ If the user is off-hook busy on an analog set and does not have ‘‘call waiting,’’ the calling endpoint hears busy. If the user is on-hook, alerting is started.

If an ASAI adjunct provides UUI in a Third Party Make Call, then the switch stores that UUI with the call for the life of the call or until overwritten due to a later UUI IE associated with the call.

Negative Acknowledgement of a User-Classified Call

A User-Classified Third Party Make Call request is denied (NAK/Return Error component) if the call fails before it is attempted in the following cases:

- The originator does not go off-hook within five seconds after originating call (and cannot be forced off-hook) (cause=CS0/18)
- The originator is out-of-service (cause=CS3/27)
- The originator is busy on one call appearance in talking state (cause=CS0/17)
- The originator has ‘‘hot line’’ administered (cause=CS0/58)

- The call could not be originated because of lack of resources (cause=CS3/40).
- A UUI Information Element longer than 32 bytes was received (cause=CS0/100).
- The originator is not a valid station extension (cause=CS0/28)
- A service circuit is requested (cause=CS3/80)
- Answering Machine Detection is requested (cause=CS3/80)

A call placed to a called endpoint whose COR does not allow the call to end will return intercept treatment to the calling endpoint and the Reorder event with cause CS0/21 (call rejected).

3. Direct-Agent Call

The direct-agent call is set up between a station user and an ACD agent logged into a specified split by using the Third Party Make Call capability.
A direct-agent call may also be placed via adjunct routing of an incoming call directly to a specified agent (see Chapter 7, ‘‘Routing Capability Group’’ for more information).

This type of call may be used by incoming call (ICM) applications whenever the application decides that the customer should talk to a specific ACD agent, not just any one in the pool. The adjunct must specify (either via table lookup or by accepting digits from the keyboard) the split extension the called endpoint ACD agent is logged into. Direct-agent calls can be tracked by Call Management Service (CMS) through the split measurements.

Valid originators for this type of call are all station extensions.

NOTE:
Another type of direct-agent call known as a Logical Direct Agent call is only available when the Expert Agent Selection (EAS) feature is enabled. See the Expert Agent Selection section of Chapter 12, ‘‘Feature Interactions’’ for detailed information.

Parameters for Direct-Agent Calls

This section contains the ASAI interface call parameters for direct-agent calls.

All parameters are the same as those listed for user-classified calls with the exception of the called endpoint.

orig_addr [mandatory] Valid calling endpoints (originators) for direct-agent calls are station extensions.

dest_addr [mandatory] Valid destinations (called endpoints) are ACD agent extensions.

split_param	Must be present and contain a valid split extension; dest_addr must be logged into this split.
priority_calling	[optional] If present, originates the call as a priority call. If not present, it defaults to nonpriority.
max_rings	N/A (Ignored)
direct_agent_call	Must be present
supervisor_assist	Must not be present
uui_info	[optional] The switch supports receiving UUI in the Third Party Make Call request from the adjunct. If an ASAI adjunct provides UUI in a Third Party Make Call, then the switch stores that UUI with the call. UUI from a Third Party Make Call will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, and in a Route Request, if one is made.
alert_order	Must not be present
service_circuit	(classifier) Must not be present
return_ack	[optional] If this parameter is present, it indicates that the switch should return an acknowledgement to the request. If this parameter is not present, then the switch does not return an acknowledgement.

Call Classification

All call-progress, audible tones are provided to the originating user (except that user does not hear dial tone or touch tones). Call progress tones are reported to the adjunct as events. For OCM preview dialing applications, final call classification is done by the station user staffing the originating set (who hears the call progress tones and manually records the result).

Switch Operation for Direct-Agent Call Setup

The switch attempts to set up the call for the extension specified in the orig_addr.

Once the call is successfully originated, the switch does not drop it regardless of outcome. The only exception is the denial outcome which results in the intercept tone being played for 30 seconds, after which the call is disconnected. The normal audible tone is provided to the originating station user. It is then up to the originating station user to drop such calls (either by going on-hook or via a third-party request). For example, if the adjunct places a call to a busy destination, the originator will be busy until he or she normally drops or until the adjunct sends a Call Control command to drop the call.

Direct-Agent Call Originator

In order for the call to be placed, the calling endpoint's voice set must have an available call appearance for origination and must not be in the talking state on any of the other call appearances. The calling endpoint is allowed to have a call on hold.

The originator may go off-hook first, and then issue the Third Party Make Call request. The switch will originate the call on the call appearance with dial tone.

If the originator is off-hook busy, the call is not placed and the request is denied (NAK/Return Error cause=CS0/17 — User busy).

If the calling endpoint is unable to originate, the switch also denies the request (cause=CS0/18 — No user responding).

If the originator is on-hook and has a speakerphone, the speakerphone is forced off-hook and the call is originated.

After dialing is completed, the calling endpoint hears call progress tones (for example, alerting or busy).

Direct-Agent Call Destination

If the destination has a display set, the display shows the specified split's name and extension.

If the destination ACD agent has a display, the display shows the name of the originator and the name of the specified split.

Once this call is set up, either the station user or the adjunct may control it. The adjunct must specify the association used in the Third Party Make Call capability when requesting other Call Control capabilities.

The switch supports receiving UUI in the Third Party Make Call request from the adjunct. If an ASAI adjunct provides UUI in a Third Party Make Call, then the switch stores that UUI with the call.

Negative Acknowledgements of a Direct-Agent Call

A Direct-Agent Third Party Make Call request is denied (NAK/Return Error component) in the following cases:

- **split_param** is not present (CS0/96)
- **split_param** option does not contain a hunt group extension (CS0/100)
- **split_param** contains an invalid hunt group extension (CS0/28)
- Originating address is not a station extension (CS0/28)
- Destination address is not a station extension (CS0/28)

- Other internal resources are unavailable (CS3/40)
- Invalid Information Element contents (CS0/100) (Either the UUI IE was longer than 32 bytes or the header was missing.)
- Service or option not subscribed/provisioned (AMD must be enabled) (CS0/50)
- The request specifies that the call is to be switch-classified or supervisor-assist as well as direct-agent. (CS3/80)
- Answering Machine Detection is requested (CS0/80)
- An EAS login ID is specified as the destination (CS0/28)

4. Supervisor-Assist Call

This call is set up via a Third Party Make Call between an ACD agent's extension and another station extension (typically a supervisor). It is measured by CMS as a supervisor-assist call. It is always a priority call.

This type of call is used by OCM and ICM applications whenever an agent (on the telephone with a client or when idle) wants to consult with the supervisor.

NOTE:
The supervisor-assist call is a feature activated through a button pushed on the ACD agent's voice set. When activating this feature from the voice set, the agent can talk to the supervisor[3] while currently on a call and then transfer the original call to the supervisor, or conference in all parties (the supervisor, the agent, and the caller).

To place a supervisor-assist call, the adjunct must specify the supervisor's extension and it must control the sequencing of events as described below. If the agent is already on a call, the original call must be put on hold[4] and then the Third Party Make Call initiated (with Supervisor-Assist option set) to the supervisor's extension (which is either provided by the agent through the data terminal or directly by the adjunct). Failure to put the first call on hold results in the switch denying the request. If the agent is not on a call, a Third Party Make Call capability with the Supervisor-Assist option is processed immediately by selection of an idle call appearance followed by call setup.

The agent must be logged into the specified ACD split to use this capability.

After talking to the supervisor, the agent may indicate (on the keyboard) that the call is to be transferred or conferenced. The adjunct translates the agent's request into a Third Party Merge request and indicates whether the agent is to be

3. The supervisor (in the non-ASAI ACD environment) is an extension defined through switch administration on the split form.
4. Either on the voice set or via a Third Party Selective Hold capability.

dropped (transfer) or not (conference) from the call. The agent may also transfer/conference this call via the voice set.

Valid originators for the ACD split specified in the request are ACD agent extensions. Agents requesting this capability must be logged in. Valid destinations are on-PBX station extensions (excluding VDNs and splits).

NOTE:
Off-PBX DCS and UDP extensions are not valid destinations.

Supervisor-Assist Call Parameters

This section contains the ASAI interface call parameters for supervisor-assist calls.

orig_addr	[mandatory] Valid calling endpoints (originators) for this type of call are ACD agent extensions.
dest_addr	[mandatory] Valid called endpoints are on-PBX station extensions (excluding VDNs, splits, off-PBX DCS and UDP extensions).
split_param	Must be present and must be a valid split extension; orig_addr should be logged into the split.
priority_calling	N/A (ignored) This call is always treated as priority for consistency with current operation.
max_rings	N/A (ignored)
direct_agent_call	Must not be present
supervisor_assist	Must be present
alert_order	Must not be present
service_circuit	Must not be present
return_ack	[optional] If this parameter is present, it indicates that the switch should return an acknowledgement to the request. If this parameter is not present, the switch does not return an acknowledgement.
uui_info	[optional] The switch supports receiving UUI in the Third Party Make Call request from the adjunct. If an ASAI adjunct provides UUI in a Third Party Make Call, then the switch stores that UUI with the call.

UUI from a Third Party Make Call will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, and in a Route Request, if one is made.

Supervisor-Assist Call Classification

A classifier is not used for this type of call. Call progress feedback is reported to the adjunct in event reports. In addition, call-progress and audible feedback is provided to the originating user.

Switch Operation for Supervisor-Assist Call Setup

The switch attempts to set up a station-to-station call for the agent's extension specified in the originating address. If the originating address is not a logged-in member of the specified split, the Denial Event Report is returned (cause=CS3/15 — Agent not logged in).

Once the call is successfully originated, the switch does not drop it regardless of outcome. The only exception is the denial outcome which results in the intercept tone being played for 30 seconds, after which the call is disconnected. It is up to either party on the call to drop (either by going on-hook or via a third-party request).

Supervisor-Assist Call Originator

In order for the call to be placed, the calling party must be logged into the specified split.

The set must have an available call appearance for origination and must not be in the talking state on any of the other call appearances. The calling party is allowed to have another call(s) on hold. The originator may go off-hook first, and then issue the Third Party Make Call request. The switch originates the call on the call appearance with dial tone.

If the calling endpoint is off-hook busy, the call is not attempted and the request is denied (NAK/return error, cause=CS0/17 — User busy).

If the calling endpoint is unable to originate, the switch denies the call request (NAK/return error, cause=CS0/18 — No user responding).

After dialing is completed, the calling endpoint hears call progress tones (for example, alerting or busy).

If the calling endpoint has a display set, the display shows the called endpoint's extension and name.

Supervisor-Assist Call Destination

Call delivery depends on the station type, station administered options, and talk state. Priority call delivery rules apply. If the destination has a display, normal display interactions for supervisor-assist calls apply. The destination does not need to be a member of the split.

Once set up, this call may be controlled either by the station user or by the adjunct. The adjunct must use the same association as that specified in the ‘‘Third Party Make Call’’ request when requesting any Third Party Call Control capabilities.

Negative Acknowledgements of a Supervisor-Assist Call

A Supervisor-Assist Third Party Make Call Request is denied (NAK/Return Error component) if:

- **split_param** is not present (CS0/96)
- **split_param** does not contain a hunt group extension (CS0/100)
- **split_param** contains an invalid hunt group extension (CS0/28)
- Originating address is not a station extension (CS0/28)
- Destination address is not a station extension (CS0/28)
- Agent not logged in (CS3/15)
- Other internal resources are unavailable (CS3/40)
- Invalid Information Element contents (CS0/100) (Either the UUI IE was longer than 32 bytes or the header was missing.)
- Service or option not subscribed/provisioned (AMD must be enabled) (CS0/50)
- The request specifies that the call is to be switch-classified or direct-agent as well as supervisor-assist. (CS3/80)
- Answering Machine Detection is requested (CS0/80)

Parameters and Call Types

To conclude this section, the following tables summarize the call parameters allowed for each of the Third Party Make Call types described above. The first table defines the four kinds of calls in terms of the allowable call parameters. The second table specifies the valid originators and destinations for a given call parameter.

Table 4-3. Third Party Make Call Options

Call Type	Orig Alert First	Dest Alert First	Service Circuit (Classif)	Direct Agent Field	Supv Assist Field	Priority Calling	Split Param.	AMD	UUI
Switch-classified	n	y	y	n	n	y	n	y	y
User-classified	y	n	n	n	n	y	n	n	y
Direct-Agent	y	n	n	y	n	y	y	n	y
Supv-Assist	y	n	n	n	y	y	y	n	y

Table 4-4. Allowable Originators and Destinations for Specific Call Options

Orig or Dest Type	Orig Alert First	Dest Alert First	Service Circuit (Classif)	Direct Agent Field	Supv Assist Field
orig=station ext	y	n	n	y	y
orig=split, hunt, annc, vdn	n	y	y	n	n
dest=station ext	y	y	x*	y	y
dest=external number	y	y	y	n	n
dest=any (int or ext)	y	y	n	n	n

*x means the option may be specified but it is not used.

Third Party Take Control

The adjunct uses this initiating capability to control a call already in progress.

Information Flow

The adjunct expects a response to its request.

An acknowledgement is sent to the controlling adjunct if the request is successful. As part of the acknowledgement, a party and extension list of all parties on the call (for all on-PBX extensions on the call) are provided.

NOTE:
The address list may contain parties in an alerting state as well as a group extension (for example, if the call is in queue and has not been delivered to an agent).

If the request is unsuccessful, the switch denies the request and sends the adjunct the appropriate cause code.

Third Party Take Control Parameters

call_id [mandatory] A call identifier for the call to be controlled. The value is typically obtained through the Event Notification or Value Query capabilities.

ACK (positive acknowledgement) Parameters

The switch responds with a list of up to six party identifiers (party_id) for the parties on the call and a list of up to six extensions of the parties on the call.

party_id list Provided by the switch and sent to the adjunct in the capability's acknowledgement. The party_id list contains the party_id for every party on the call. Each party identifier corresponds to an entry in the extension list.

extension list [optional] Specifies the extension address of each on-PBX party to the call. Addresses of every on-PBX party on the call are provided for alerting, connected, or queued parties. The address for queued parties is the split extension of the queue. The default station extension is provided for off-PBX parties.

Denial (NAK) Causes

The switch issues the following reason for denying the request:

- Invalid association (CS0/81)
  The association is already in existence.
- Requested facility not subscribed/provisioned (CS0/50)
  The user has not subscribed for the requested capability.
- Invalid call_id (CS3/86)
  The requester has sent an invalid call_id (call does not exist, has been cleared).
- Invalid Information Element contents (CS0/100)
  The call_id is outside the range of the maximum call_id value.
- Service or option not available (CS3/63)
  The call is already being controlled.
- Mandatory Information Element missing (CS0/96)
  A required parameter is missing in the request.
- Resources not available (CS3/40)
  The maximum number of calls being controlled has been exceeded.
- Switching equipment congestion (CS0/42)
  The switch is not accepting the request at this time because of traffic congestion. The adjunct or user may wish to retry the request but should not do so immediately.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The switch allows only one adjunct to control a call at any given time. Subsequent requests for Third Party Take Control from an adjunct while a call is already active are denied by the switch. Third Party Take Control cannot take control of a call which has been established over, and is still controlled over, a Call Control association.

Third Party Take Control may be issued at any time during the life of a call.

CAUTION:
Alerting parties are reported as ''Connected'' Parties in the response to Third Party Take Control.

NOTE:
It is possible to have only one party reported (for example, the call is in the process of being adjunct-routed).

Third Party Selective Hold

The adjunct uses this capability to place a controlled call on hold at an on-PBX station. The effect is as if the specified party depressed the hold button on his or her multifunction terminal to locally place the call on hold, or flashed the switch hook on an analog terminal.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Places the call on hold and sends the adjunct an acknowledgement, or

— Denies the request

Third Party Selective Hold Parameters

party_id Identifies the party to be placed on hold. This party must be in the active (talking) state or already held.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for denying the request:

- Invalid association (CS0/81)
  The association does not exist.
- Invalid number/domain (CS0/28)
  The party_id given is invalid or does not correspond to a station.
- Message not compatible with call state (CS0/98)
  The request is not compatible with the call state. The party to be put on hold is not currently active (talking), so it cannot be put on hold. Also applies to analog sets with two calls already on hold.
- Mandatory Information Element missing (CS0/96)
  A required parameter is missing in this request.
- Invalid Information Element contents (CS0/100)
  The party_id or call_id value of the request is invalid. For example, a Third Party Selective Hold is sent for the destination of a ringing call (the call is ringing and not yet answered at that endpoint). The party_id specified in the party_id Information Element has not passed through the connected stage and is therefore invalid.

- User busy (CS0/17)
  The user is busy with another ASAI request.
- Invalid call type (CS3/43)
  The call cannot be held due to the type of call (for example, emergency, wakeup, or service observed).

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The adjunct must know the party_id before placing the party on hold. A party may only be put on hold if it is in on the call.

After a party is placed on hold through a Third Party Hold request, the user does not receive dial tone (regardless of the type of phone [set]). Thus, subsequent calls must be placed by selecting an idle call appearance or through the Third Party Make Call Request.

Only station extensions support this capability; any requests containing party_ids corresponding to a trunk will be denied.

If the party is already on hold on the specified call when the switch receives the request, a positive ACK is returned.

NOTE:
An analog set can be looked at as having two call appearances, CA1 and CA2.

1. If the hold request is for either call appearance already in the held state, nothing else is done (request is ACKed).
2. If the hold request is for CA1 (when CA1 is in the active state), the call is placed on soft (conference) hold.
3. If the hold request is for CA2 (when CA2 is in the active state), and CA1 is also held, then CA2 is placed on hard-hold.
4. If the hold request is for CA2 (when CA2 is in the active state), and CA1 is idle, the call is moved to CA1, and the call is placed on soft (conference) hold.

5. In cases where the request for hold is for CA2, but CA1 already has a held call, and the call waiting is active, then the hold request is denied.

When you set up repeated conferences from an analog set, you start with the call on CA1, put it on hold (rule #2), and conference, the call ends up on CA2. The next hold request will find the call on CA2, with CA1 idle (rule #4). Each time you complete the conference, your call remains on CA2, so rule #4 applies repeatedly.

Third Party Reconnect

The adjunct uses this controlling capability to reconnect an on-PBX party to a controlled call.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Reconnects the party to the call after validating the parameters, or

— Denies the request

Third Party Reconnect Parameters

party_id	Identifies the party to be reconnected

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for denying the request:

- Invalid association (CS0/81)
  The association does not exist.
- Invalid number/domain (CS0/28)
  The party_id is incorrect or invalid.
- Invalid Information Element contents (CS0/100)
  The party_id value of the request is invalid. For example, a Third Party Reconnect is sent for the destination of a ringing call (the call is ringing and not yet answered at that endpoint). The party_id specified in the party_id Information Element has not passed through the connected stage and is therefore invalid.
- Message not compatible with call state (CS0/98)
  The party is not currently in the ‘‘hold’’ state so it cannot be reconnected. The party was on-hook and did not go off-hook within five seconds after the request was made (the call remains on hold).
- Resources not available (CS3/40)
  The host attempted to connect a seventh party to a six-party conference call.

- Mandatory Information Element missing (CS0/96)
  The party_id is missing in the request.
- User busy (CS0/17)
  The user is busy (active) on another call.
- User not responding (CS0/18)
  The party was on-hook when the request was made and it did not go off-hook within five seconds (call remains on hold).
- Resources not available (CS3/43)
  The call cannot be reconnected due to too many parties already on the call or other switch-specific resource problems.

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The adjunct uses previously received information about the party to release it from the held state. This party must have been placed on (hard or soft) hold from the station set or via the adjunct. A party may be reconnected only to the same call from which it had been put on hold as long as there is no other active call at the user's station. If the party is already reconnected on the specified call when the switch receives the request, a positive ACK is returned.

If the user is on-hook (in the held state), the switch must be able to force the station off-hook or the user must go off-hook within five seconds after requesting a Third Party Reconnect. If one of the above conditions is not met, the request is denied and the party remains held.

If the user is listening to dial tone while a request for Third Party Reconnect is received, the dial tone is dropped and the user is reconnected to the held call.

If the user is listening to any other kind of tone (for example, denial) or is busy talking on another call, the Third Party Reconnect request is denied (CS0/17).

Redirect Call

The adjunct uses this capability to re-route a call that is already alerting a station extension.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Redirects the call and drops the alerting party after validating the parameters, or

— Denies the request and maintains the alerting party

Third Party Redirect Call Parameters

redirecting party_id — Identifies the party to be redirected from (must map to a valid voice station extension, which includes off-premises stations ([OPS]). If not specified, call is redirected away from ‘‘last added party’’ to a call.

redirected-to-number — [mandatory] Identifies the number the call is routed to (may be on- or off-switch). If more than 24 digits are provided, only the first 24 are used.

ACK (Positive Acknowledgement) Parameters

An acknowledgment is sent to the adjunct if the request is successfully completed and the call stops alerting the redirecting station.

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for denying the request:

- Request is for release prior to G3V4 (CS0/111)
  This service is only available beginning with release G3V4.
- Mandatory Information Element missing (CS0/96)
  The redirected-to number or other IE is missing in the request.
- Invalid CRV (CS0/81)
  The Call Reference Value is missing or invalid.
- Invalid destination (CS3/43)
  Invalid destinations include the following: empty (0 digits), unassigned extension (invalid dial plan numbers), incomplete number of digits for AAR/ARS pattern, or non-AAR/ARS FAC.

- Redirected-to Station Busy, or Terminating Extension Group (TEG) has one or more members busy (CS0/17)
- Miscellaneous Restrictions (CS3/42)
  - The redirected-to number cannot be the same as the originating number or the redirecting number.
  - The call is redirecting on the first leg of a switch-classified call.
- Miscellaneous Restrictions (CS3/43)
  - The redirected-to number is a Remote Access Extension, or the COR check fails.
  - The redirecting station is origination-restricted.
- Miscellaneous Restrictions (CS3/63)
  - The redirecting number is not a station extension, the call_id does not exist, or the call is not in the alerting state or is redirecting while in vector processing.
  - Calls that result in intercept treatment will not be allowed to redirect even if normally such calls would be sent to an attendant.

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

With this service, when a call is routed to a new number, if the new number is a station on the switch, it begins alerting (if available to receive the call) and the originally-alerted party is dropped. At this point, this alerting call is eligible to be redirected again. If the new (redirected-to) number is off-premises, the call is routed there and the caller hears call progress tones. In this case, the call may not be redirected again.

For both on- and off-switch redirection, the switch will not drop the redirecting party until success is assured. It is at this point that the positive acknowledgement is sent. If the switch cannot redirect the call, NAK is provided and the alerting call will not be affected.

There is a special case where an adjunct can receive a positive acknowledgment, but the redirect may fail. For example, if the call is redirected to an off-premises number and the network trunk is seized, the switch considers this successful redirection and drops the redirecting party. The caller may hear any network-provided call progress tones.

Party_ids may be re-used if the call is redirected more than once.

An application using this feature should consider the timing for redirecting a call, particularly when redirecting away from a display user. If a display station user gets an incoming call and the application redirects it immediately, the station user may not have enough time to read the display (the incoming call information) before it is cleared by the application.

Since it is not possible to redirect busy calls, if possible an application should check whether the call had resulted in a busy condition before attempting to redirect it. This may be done if the application receives the appropriate call progress events from the switch (such as ‘‘busy,’’ ‘‘reorder/denial,’’ or ‘‘connected’’). ASAI Redirect Alerting Call is not disabled when trunk-to-trunk transfer is disabled.

The call state may change between the Alerting event and the service request for redirection. An application should therefore expect certain NAKs and handle them appropriately.

Third Party Merge

The adjunct uses this controlling capability to merge (for example, conference or transfer at a specified station) two existing controlled calls into a single call on behalf of an on-PBX station. A request is made over the association of the call on hold.

NOTE:
This only works if one call is on hold while the other is active (alerting, queued, held, or connected).

Information Flow

The adjunct expects a response to its request.

The switch either:

— Merges the two calls after it has validated the parameters, or

— Denies the request if the calls and specified station are not specified or are not in the correct state

Third Party Merge Parameters

common_party_id	[mandatory] Identifies the common endpoint with regard to the held call. This station must have a call on hold and one active (talking) state call.
call_id	[mandatory] The call_id of the active call.
conf/trans_flag	The type of merge (Conference/Transfer) is set with this parameter. The conf/trans_flag must be PRESENT. If the flag is set to TRANSFER, the calls are merged and the common party is dropped from the call. The transfer operation merges the two existing calls and drops the common party from the call. If the conf/trans_flag is set to CONFERENCE, the calls are merged together and the common party is retained in the resultant call.

ACK (Positive Acknowledgement) Parameters

The switch replies with a call identifier for the merged call (call_id), a list of up to six party identifiers for the parties on the call (party_id), and a list of up to six extensions of the parties on the call.

An acknowledgement (with party/extension information) is sent to the controlling adjunct when the conference/transfer is complete. Included in the acknowledgement is a list of all parties on the call (including extensions for local parties).

The association used to request the capability is used for the acknowledgement.

Any other associations monitoring the call or Domain Controlling endpoints receive the Call Conference/Call Transferred Event Reports.

result_call_id The call_id after the call is merged (provided by the switch)

party_id list The list of party_ids for the merged call (provided by the switch)

extension list The list of extension numbers for the parties on the merged call. Off-PBX parties always provide the default station extension. The address for queued parties is the split extension of the queue.

Denial (NAK) Cause

The switch issues one of the following reasons as the cause for denying the request:

- Invalid association (CS0/81)
  The association is nonexistent.
- Invalid number/domain (CS0/28)
  The controlling party has not been specified correctly.
- Message not compatible with call state (CS0/98)
  The common party is not in a valid state for the operation (merge) to take place. For example, the common party does not have one call active (talking) and one call in the held state as required.
- Resources not available (CS3/40)
  The switch may have run out of resources (for example, time slots).

- Reorder/Denial (CS3/42)
  Both calls are Alerting.

  Both calls are being service observed.

  An active call is in a vector processing stage.

  The host attempted to add a seventh party to an existing six-party conference call.
- Invalid Information Element contents (CS0/100)
  The party_id value of the request is invalid.
- Mandatory Information Element missing (CS0/96)
  The party_id or call_id is missing from the request.

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

Both calls must be controlled (for example, Call Control associations) for Third Party Merge to operate successfully.

For analog sets (phones), Third Party Merge is only allowed if one call is held and the second is active (talking). Calls on hard hold or alerting cannot be affected by a Third Party Merge request.

After the merge, the station is off-hook idle.

The Third Party Merge capability can be also used by the adjunct to transfer an ACD call to a supervisor. A sample scenario is presented below. Note that this sample is not the only way in which to effect the transfer to the supervisor.

An ACD agent handling may request (via data keyboard) that the call be transferred to the supervisor. The adjunct:

1. Places the existing call on hold.
2. Issues a Third Party Make Call request (with the supervisor-assist option) to the supervisor's extension.

3. When the supervisor either is alerted or answers, the adjunct may merge the supervisor call and the previously held call using the Third Party Merge capability. (This capability is requested on the same association as the held call).

Since this was a transfer request, the conf/trans_flag parameter is set to TRANSFER and the switch drops the ACD agent from the connection after the merge takes place.

If the ACD agent requested a supervisor conference, the same procedure would have taken place but the conf/trans_flag would have been set to CONFERENCE; the switch does not drop the ACD agent from the resulting connection. Thus, a three-party call is created.

Third Party Selective Drop

The adjunct uses this controlling capability to disconnect a specific party from a call. The party may be a station or a trunk. For G3V3 and later, a tone resource (other than ringback) may be dropped from a connection.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Drops the party once it has validated the parameters, or

— Denies the request if it could not execute the disconnect operation or if the party_id specified does not exist (invalid parameter)

Third Party Selective Drop Parameters

party_id [mandatory] Identifies the party on the call to be disconnected.

uui_info If there is UUI stored with a call, and if that UUI came from an ISDN DISCONNECT message or in an ASAI Third Party Drop Request, and if a party drops from the call and a Drop Event Report results, then the switch will include the UUI stored with the call in the Drop Event Report.

UUI received in a successful Third Party Selective Drop on the ASAI link is stored by the switch with the call for the life of the call or until overwritten due to a later UUI IE associated with the call.

resource_id [optional] Identifies a tone resource to be dropped. If specified, do not specify party_id.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for denying the request:

- Mandatory Information Element missing (CS0/96)
  The party_id is missing from the request.

- Invalid number/domain (CS0/28)
  An invalid party_id or extension number is present in the request.
- Invalid association (CS0/81)
  The association does not exist.
- Both party_id and resource_id are specified (CS0/80)
- Resource_id specifies a tone resource and one is not active on the connection, or the active tone is ringback. (CS0/82)
- A drop that is not allowed was requested (CS3/12)
  A drop was requested while the trunk was in the wrong state (for example, prior to finishing dialing).
- Message not compatible with call state (CS0/98)
  The call is not currently active or is in a hold state and therefore cannot be dropped.
- Invalid Information Element contents (CS0/100)
  The party_id value is invalid; for example, it is out of range.

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The adjunct must know the association and the party_id of the party on the call to be dropped. When a party is dropped from an existing conference call with three or more parties (directly connected to the switch), the other parties remain on the call. If this is a two-party call, the entire call is dismantled.

Only *connected* parties and trunks (in any state) can be dropped from a call. Held, bridged, and alerting local parties cannot be dropped by the adjunct.

Third Party Selective Drop cannot be used to drop parties from a switch-classified call while the call is in the process of being classified. Third Party Clear Call should be used instead.

Third Party Selective Drop may not be used to drop a call/party out of queue. Third Party Selective Drop may not be used with resource_id if the tone provided is not a local tone on the switch (for example, a network-provided tone).

Third Party Relinquish Control

The adjunct uses this capability to terminate a Call Control association. The call itself is not affected by this capability.

Information Flow

The adjunct expects a response to its request. The switch:

— Relinquishes control of the call if the request is valid

Third Party Relinquish Control Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause

This request receives a NAK for a switch-classified call if the request is made while the call is being classified (CS0/98).

Protocol Error (NAK) Cause

None for this capability

Considerations

After an adjunct has invoked Third Party Relinquish Control for a call, the same adjunct or another adjunct can take control of the same call by invoking the Third Party Take Control capability.

If this call is being monitored (Event Notification active), the adjunct continues to receive event reports over the event notification association.

This operation does not disconnect any parties from the call; switch call processing continues for the duration of the call.

Third Party Clear Call

The adjunct uses this controlling capability to drop all parties from a controlled call.

Information Flow

The adjunct expects a response to its request.

The switch either:

- — Clears the entire call (for example, disconnects all parties) and acknowledges, or
- — Denies the request

As the switch is clearing the call:

- Every station dropped is in the off-hook idle state.
- Any lamps associated with the call are off.
- The displays are cleared.

Third Party Clear Call Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause

The switch issues the following reason as the cause for denying the request and ending the association:

- Invalid association (CS0/81)
  The association does not exist.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The Third Party Clear Call capability can only clear calls controlled by the Call Control capabilities.

Third Party Send DTMF Signals (G3V4)

The adjunct uses this controlling capability to transmit a sequence of DTMF tones on behalf of a party on the call.

Information Flow

The adjunct expects a response to its request.

The switch either:

- — Transmits the tones on the sender's talk path after it has validated the parameters, or
- — Denies the request if it is unable to service it

Third Party Send DTMF Signals Parameters

tx_party_id [mandatory] Identifies the party on whose behalf the DTMF signals are to be sent. This party must map to a valid, physical endpoint on the call.

rx_party_id [optional] If present, this parameter is ignored. A list of up to 5 party_ids that will receive the DTMF signals. If this list is null, then all parties on the call receive the DTMF signals if otherwise eligible (that is, if they are connected via ports that support end-to-end signaling). See ''Considerations.''

char_seq [mandatory] Identifies the character sequence to be generated (maximum 32).

tone_dur [optional] If present, this parameter is ignored.

pause_dur [optional] If present, this parameter is ignored.

ACK (Positive Acknowledgement) Parameters

An acknowledgement is sent when it is determined that the DTMF digits could be transmitted, not when done transmitting. There are no parameters contained in this ACK.

Denial (NAK) Cause

The switch issues the following reasons as the cause for denying the request and ending the association:

- Mandatory Information Element missing (CS0/96)
  The char_seq is missing from the request. A denial may be sent if another parameter is missing as well.
- Invalid number/domain (CS0/28)
  Party_id is out of range or is a party o whose behalf ASAI cannot send DTMF tones (that is, a party other than a station or a trunk). Or, call_id is out of range.
- Invalid association (CS0/81)
  The request is supported only over call control and domain control associations. An attempt has been made to request this service over another type of association.
- Invalid Information Element contents (CS0/100)
  A value of an IE is outside the range specified, or a character sequence with a length of 0 or invalid characters has been supplied.
- Service or Option Not Available (CS3/63)
  The provided call ID does not exist.
- Message Not Compatible with Call State (due to call state) (CS3/98)
  DTMF signals can be generated only for active calls on which no other signaling tones are present. Or, the switch capacity for simultaneous sending of DTMF digits is full. Also, the call must not be in vector processing or in a non-active state. The switch has detected that local, audible signals, including (PBX) dial tone, busy tone, ringback tone, intercept tone, or Music-on-Hold/Delay are currently being received.
- Reorder/Denial (CS3/42)
  A request to send DTMF on a conference call with more than 5 parties is denied.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

- The Send DTMF Signals feature conforms to the CSTA Send DTMF Digits service specification in CSTA Version II.
- DTMF signals may be sent to any extension type on the call.
- If a character sequence with a length greater than 32 is supplied, only the first 32 characters in the sequence will be accepted (the remaining characters will be ignored).
- The switch does not provide any signal to the adjunct when transmission has completed.
- Each digit sent will require 350 ms. Interdigit timing is 100 ms.
- While DTMF signals are being transmitted, all parties on the call will be disconnected from the talk path.
- If a party drops from the call while tones are being sent and others remain on the call, signals will continue to be sent to the remaining parties.
- If a party is added to a call while signals are being sent, this party will hear the remaining digits but will not be talk path disconnected. As a result, clicking noises may be heard.

Third Party Call Ended

The switch uses this terminating capability to inform the adjunct that a controlled call has ended and to clear the call control association that was controlling the call. It is also used when two calls are merged since one call control association terminates when the calls merge.

Information Flow

The switch does not expect a response.

Third Party Call Ended Parameters

call_id Identifies the call that has terminated

Causes

The switch issues one of the following reasons as the cause for ending the association.

- Normal clearing (CS0/16)
  The call having the requested call identity has been cleared. The call is no longer active. The extraneous call is dropped as the result of merging the call.
- User busy (CS0/17)
  The first alerted party on the call is busy and the call is dropped.
- Classifiers not available (CS3/21)
  - The call is dropped because no call classifiers are available.
  - COR check failed.
- Call with requested identity has been terminated (CS3/86)
  Two calls have been merged into one.
- Trunks not available (CS3/20)
  The call is dropped because no trunks are available.
- Split Queues full (CS3/22)
  The call is dropped because it cannot be queued.
- Denial/Reorder (CS0/42)
  The reorder tone is detected.
- Outgoing calls barred (CS0/52)
  The calling party number of a switch-classified call has an FRL lower than that of the trunk group.
- Call rejected (CS0/21)
  The switch drops the call due to an illegal action and/or request by the user (for example, attempting to merge two outbound calls without disconnect supervision).

- No answer (CS3/19)
  The classifier does not detect answer within the allowed number of rings.
- Answering Machine Detected (CS3/24)
  The call was answered by an answering machine.
- Incompatible Destination (CS0/88)
  The classifier detects answer back tone.
- No Circuit or Channel Available (CS0/34)
- Unassigned Number (CS0/01)
- Invalid Number/Domain (CS0/28)
- Normal Unspecified/SIT — Vacant (CS0/31)
- Normal, Unspecified (CS0/127)
  This value is a normal drop cause value when something other than an ISDN endpoint drops out of a connection. For example, when the host issues a Third Party Make Call (User-Classified) from a station to an announcement extension and the call is successfully completed. The calling party hears the announcement and the call is dropped. The Disconnect/Drop Event Report is also generated. See Chapter 3, ''Common Capabilities,'' for more information on the Disconnect/Drop Event Report.
- Denial/Reorder (CS3/42)
  Provided when a switch-clasified call (that has been answered by the far end) cannot be transferred to the intended split/VDN.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Cause

None for this capability

ASAI and Domain Control

5

This chapter describes the Domain (Station or ACD Split/EAS skill) Call Control Capability Group. The adjunct uses these capabilities to control and monitor all calls at a station extension, to monitor selected calls only at that specific station, and to begin outbound calls from that station. Additionally, the adjunct may use this capability to receive Agent Logout Event Reports, and, starting with G3V4, Agent Login Event Reports, for a specified ACD split or EAS skill.

Domain control capabilities are as follows:

Third Party Domain Control Request for Station Domain
When used with a station domain, the adjunct uses this initiating capability to ask the switch for event reports on calls at a specified domain (station extension).

Third Party Domain Control Request for ACD Split/EAS Skill Domain
An adjunct may use this capability to receive Agent Logout Event Reports and, starting with G3V4, Agent Login Event Reports for a specified ACD split or EAS skill.

Third Party Answer
This capability allows an application to make a request on behalf of a station user to ‘‘answer’’ a ringing, bridged, or held call present at a station.

Third Party Selective Hold
This controlling capability lets the adjunct put a specified call on hold at the extension being controlled.

Third Party Reconnect	The adjunct uses this controlling capability to reconnect a specified call at a controlled extension.
Redirect Call	The adjunct uses this capability to re-route a call that is already alerting a station extension.
Third Party Merge	The adjunct uses this controlling capability to merge two calls at a controlled extension.
Third Party Selective Drop	This controlling capability lets the adjunct drop a controlled extension from a given call.
Third Party Auto Dial	The adjunct uses this controlling capability to begin an outbound call on behalf of a station extension.
Third Party Relinquish Control	The adjunct uses this controlling capability to end an active domain-control association.
Third Party Send DTMF Signals	The adjunct uses this controlling capability to transmit a sequence of DTMF tones on behalf of the domain-controlled party on a specified call.
Third Party Domain Control Ended	The switch uses this terminating capability when a domain-control association ends.
Event Reports	See Chapter 3, ''Common Capabilities.''
U-Abort (User Abort)	See Chapter 3, ''Common Capabilities.''

NOTE:
As previously stated, this capability group is actually a subsection of the ''Call Control Capability Group'' section in Chapter 4, but it is treated as a separate capability group in this chapter.

Domain (Station) Control Description

Domain (station) Control allows an adjunct to receive event reports and control all calls beginning at or coming to a specific station extension. Without these capabilities, similar control and monitoring functions would require the adjunct to have adjunct control of the entire call. It allows the adjunct to control only the station extension associated with the Domain Control association instead of allowing control of all parties (extensions) on a call.

When a call leaves the extension domain, event reports for the call over the domain-control association cease.

Station Domain

A station domain is a valid station as specified in the dialing plan for which capabilities in the Domain Control subset can be requested. Station domains are limited to voice stations (including administered extensions without hardware) that are locally connected to the switch. It excludes, for example, the following:

- Data extensions
- Attendant console extensions
- Announcement extensions
- Off-PBX DCS/UDP extensions
- Any group-type extensions (hunt groups, ACD split or EAS skill extensions)
- VDNs

One station domain maps to a physical set identified by a unique primary extension number (the number an administrator would give as a parameter in the **display station** command to display information for the station set). One station domain encompasses all call appearances at the physical station set, including all primary, bridged (from other primary extensions), TEG, and PCOL call appearances at the physical station set.

Split Domain

A split domain is a valid ACD split or EAS skill extension specified on the Hunt Group form. Only the Agent Logout Event Report, and, starting with G3V4, the Agent Login Event Report, are provided for this domain. Third Party Call Control capabilities are not available for this domain.

Domain Call Control Capabilities

The following capabilities have the same function within Domain Call Control as they do in the Call Control Capability Group.

- Third Party Merge
- Third Party Selective Hold
- Third Party Reconnect
- Redirect Call
- Third Party Selective Drop

The following parameters allow the use of these capabilities over the Domain Control Capability Group.

Call_id A call_id parameter is included in these capabilities for their use within a Domain Control association. Because a Domain Control association may control a number of calls, this parameter selects the call affected. An adjunct must always specify the call_id when requesting a Call Control Capability over a domain control association.

The following table summarizes whether a specific Call Control request is or is not allowed based on the specified party state (bridge with multifunction station set principal is assumed and no exclusion feature active).

Table 5-1. Call Control Acceptance in Various Party States

Domain Control Request	Active St	Held St	Alerting St	Idle St	Dialing St	Bridged St
Third Party Auto Dial	no[1]	yes[2]	yes[2]	yes[3]	yes[4]	yes[3]
Third Party Answer	yes	yes	yes	no	no	yes
Third Party Selective Drop	yes	no	no	no	yes	no
Third Party Selective Hold	yes	yes	no	no	no	no
Third Party Reconnect	yes	yes	no	no	no	no
Third Party Merge(ctling pty)	yes	yes	no	no	no	no
Redirect Call	no	no	yes	no	no	no

[1] User active on any call appearance or call
[2] On a call appearance other than the one in the given state
[3] User with at least one idle primary call appearance
[4] Call appearance selected for call origination

Third Party Domain (Station) Control Request

An adjunct uses this capability to receive event reports and control calls at a specified extension.

Information Flow

The adjunct expects a response to its request.

The switch returns an acknowledgement to the application after the switch verifies that the station number is valid and the maximum number of controllers for the station has not been reached.

Domain (Station) Control Request Parameters

domain [mandatory] This parameter identifies a valid local station.

ACK (Positive Acknowledgement) Parameters

The switch responds to the adjunct's request with a list of call_ids, the party_id of the principal's extension on the call, and the state of the principal's extension on the call.

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid association (CS0/81)
  The association does not exist.
- Invalid number/domain (CS0/28)
  An invalid address or extension number is present in the request.
- Service or Option Not Available (CS3/63)
  The maximum number of Domain Control Requests are in use for the domain.
- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of processor overload. The adjunct or user may wish to retry the request but should not do so immediately.

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Domain Control request on a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221.

Third Party Domain Control Request for ACD Split/EAS Skill Domain

This capability allows the adjunct to receive event reports at a specified domain. Currently, only the Logout Event Report, and, starting with G3V4, the Login Event Report, are available. Third Party Call Control capabilities are not allowed for Third Party Domain Control for the ACD Split/EAS Skill Domain.

Information Flow

The adjunct expects a response to its request.

The switch returns an acknowledgement to the application after the switch verifies that the ACD split or EAS skill is valid.

Domain Control Request for ACD Split/EAS Skill Domain Parameter

domain [mandatory] This parameter identifies the ACD split or EAS skill.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid Association (CS0/81)
  The association is already in use.
- Invalid Number/Domain (CS0/28)
  The number contained in the request is an invalid parameter value.
- Service or Option Not Available (CS3/63)
  The maximum number of Domain Control requests are in use for the domain.
- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of processor overload. The adjunct or user may wish to retry the request but should not do so immediately.

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The ‘‘logout’’ event is reported for split domain-controlled associations when an agent logs out of the split. This capability remains in effect for the split domain until either the adjunct invokes the Third Party Relinquish Control capability, or the switch invokes the Third Party Domain Control Ended capability. Call control and call-related events are not reported with this capability.

Starting with G3V4, the ‘‘login’’ event is reported for split domain-controlled associations when an agent logs into the split. It is provided regardless of how the agent logs in (either manually through FACs or through ASAI).

Third Party Answer

This capability allows the adjunct to request, on behalf of a station user, that a ringing, bridged, or held call present at a station be ‘‘answered.’’ This is done by connecting a call by forcing the station off-hook, if the user is on-hook, or by cutting through the call to the head or handset, if the user is off-hook (listening to dial tone or being in the off-hook idle state). The effect is as if the station user selected the call appearance of the alerting, bridged, or held call and went off-hook.

Information Flow

The adjunct expects a response to its request.

The Third Party Answer request is acknowledged (ACK) by the switch if the switch is able to connect the specified call by either forcing the station off-hook (turning the speakerphone on) or waiting up to five seconds for the user to become off-hook.

Third Party Answer Parameters

party_id [ignored] Indicates the alerting, bridged, or held party on the call that must be connected to the call.

call_id [mandatory] Indicates the alerting, bridged, or held call that must be connected at the controlled station.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid Association (CS0/81)
  The association does not exist.
- User Busy (CS0/17)
  The station is busy on a call or there are no idle call appearances available.
- No User Responding (CS0/18)
  The station user did not go off-hook within five seconds and is not capable of being forced off-hook.

- Message Not Compatible with Call State (CS0/98)
  The specified call at the station is not in the alerting, bridged, held state or active state.
- Resources not available (CS3/40)
  The host attempted to add a seventh party to a call with six parties active (bridging and held cases only).
- Invalid number/domain (CS0/28)
  The call_id contained in the request is invalid.
- Mandatory Information Element missing (CS0/96)
  A mandatory parameter is missing from the request.

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The Third Party Answer capability can be used to answer a call present at any station type (for example, analog, DCP, hybrid, and BRI), as long as the station is domain-controlled. A call that is already connected when the Third Party Answer request is made results in a positive acknowledgement (ACK).

Multifunction Station Operation

For a multifunction station user, this capability is successful in the following cases:

- The user is being alerted on-hook and can either be forced off-hook or is manually taken off-hook within five seconds of the request (the switch selects the ringing call appearance).
- The user is off-hook idle; the switch selects the alerting call appearance and answers the call.
- The user is off-hook listening to dial tone; the switch drops the dial tone call appearance and answers the alerting call on the alerting call appearance.

A held call is answered on the held call appearance, providing the user is not busy on another call.

A bridged call is answered on the bridged call appearance, providing the user is not busy on another call, or the exclusion feature is not active for the call.

An Automatic Callback (ACB), PCOL, or TEG call is answered on a free call appearance, providing the user is not busy on another call.

A multifunction station user can also have other call appearances with alerting, bridged, or held calls while requesting the Third Party Answer capability on a call.

If the station is active on a call (talking), listening to reorder/intercept tone, or does not have an idle call appearance (for ACB, ICOM, PCOL, or TEG calls) at the time the Third Party Answer capability is requested, the request is denied.

Analog Station Operation

For an analog station user, this capability is successful only under the following circumstances:

- The user is being alerted on-hook (and is manually taken off-hook within five seconds).
- The user is off-hook idle (or listening to dial tone) with a call waiting; the switch drops the dial tone (if any) and answers the call waiting call.
- The user is off-hook idle (or listening to dial tone) with a held call (soft or hard); the switch drops the dial tone (if any) and answers the specified held call (there could be two held calls at the set, one soft-held and one hard-held).

An analog set may only have one or two held calls when invoking the Third Party Answer capability on a call. If there are two held calls, one is soft-held, the other hard-held. Third Party Answer of any held call (in the absence of another held call and with an off-hook station) resets the switch-hook flash counter to zero, as if the user had manually gone on-hook and answered the alerting/held call. Third Party Answer of a hard-held call (in the presence of another, soft held call, and with an off-hook station), leaves the switch-hook flash counter unchanged. Thus, the user may use subsequent switch-hook flashes to effect a conference operation between the previously soft-held call and the active call (reconnected from hard-hold). Third Party Answer of a hard-held call in the presence of another soft-held call and with the station on-hook is denied. This is consistent with manual operation because when the user goes on-hook with two held calls, one soft- and one hard-held, the user is re-alerted and goes off-hook, and the soft-held call is retrieved.

If the station is active on a call (talking) or listening to reorder/intercept tone at the time the Third Party Answer capability is requested, the request is denied (CS0/17 — User Busy).

Third Party Selective Hold

The adjunct uses this capability to place a controlled extension on hold. The effect is as if the specified party depressed the hold button on his or her terminal to locally place the call on hold. For analog sets (phones) with only one call active, selective hold places the call on conference hold (the same as if the switch-hook was flashed once). For analog sets which already have a held call, this request places the active call on hard hold.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Places the call on hold and sends the adjunct an acknowledgement, or

— Denies the request (see causes below)

Third Party Selective Hold Parameter

call_id	[mandatory]	Identifies the call to be placed on hold

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid association (CS0/81)
  The association does not exist.
- Invalid number/domain (CS0/28)
  The number contained in the request is an invalid call_id value.
- Invalid state (CS0/98)
  The call is not in the talking state in order to be put on hold.
- Mandatory IE missing (CS0/96)
  The required call_id is missing in this request.
- Invalid information element contents (CS0/100)
  The party_id or call_id value of the request is invalid (for example, the party is alerting).
- User Busy (CS0/17)
  The user is busy with another ASAI request.

- Invalid call type (CS3/43)
  The call cannot be held due to the type of call (for example, emergency, wakeup, or service observed).

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The adjunct must know the call_id before placing the extension on hold.

If the call is already on hold at the controlled station, when the switch receives the request, a positive acknowledgement (ACK) is returned. See the ‘‘Third Party Reconnect’’ section in Chapter 4 for additional ‘‘Considerations.’’

Third Party Reconnect

The adjunct uses this controlling capability to reconnect a held call to a station.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Reconnects the call after validating the parameters, or

— Denies the request if the call_id parameter is incorrect

Third Party Reconnect Parameters

call_id	[mandatory]	Identifies the call to be reconnected

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid association (CS0/81)
  The association does not exist.
- Invalid number/domain (CS0/28)
  The call_id is incorrect or invalid.
- Invalid state (CS0/98)
  The call is not currently in the hold state so it cannot be reconnected.
- Mandatory Information Element missing (CS0/96)
  The call_id is missing in the request.
- Resources not available (CS3/43)
  The call cannot be reconnected due to too many parties on the call or other switch-specific resource problems.
- No User Responding (CS0/18)
  The party was on-hook when the request was made and it did not go off-hook within five seconds (call remains held).

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The adjunct uses previously acquired call_id information about the call to release it from the held state. The call must have been placed on (hard or soft) hold from the station set or via the adjunct.

If the party is already reconnected on the specified call when the switch receives the request, an ACK is returned.

Redirect Call

The adjunct uses this capability to re-route a call that is already alerting a station extension.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Redirects the call and drops the alerting party after validating the parameters, or

— Denies the request and maintains the alerting party

Domain Control Redirect Call Parameters

call_id	[mandatory] The call_id of the call to be redirected
redirected-to-number	[mandatory] Identifies the number the call is routed to (may be on or off-switch). If more than 24 digits are provided, only the first 24 are used.

ACK (Positive Acknowledgement) Parameters

An acknowledgment is sent to the adjunct if the request is successfully completed and the call stops alerting the redirecting station.

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for denying the request:

- Request is for release prior to G3V4 (CS0/111)
  This service is only available beginning with Release G3V4.
- Mandatory Information Element missing (CS0/96)
  The call_id or redirected-to number or other IE is missing in the request.
- Invalid CRV (CS0/81)
  The Call Reference Value is missing or invalid.
- Invalid destination (CS3/43)
  Invalid destinations include the following: empty (0 digits), unassigned extension (invalid dial plan numbers), incomplete number of digits for AAR/ARS pattern, or non-AAR/ARS FAC.
- Redirected-to Station Busy, or Terminating Extension Group (TEG) has one or more members busy (CS0/17)

- Miscellaneous Restrictions — Redirected-to Number
  The redirected-to number is a Remote Access Extension, or the COR check fails (CS3/43).
  The redirected-to number cannot be the same as the originating number or the redirecting number (CS3/42).
- Miscellaneous Restrictions — Redirecting Number
  - The redirecting number is not a station extension, the call_id does not exist, or the call is not in the alerting state or is redirecting while in vector processing (CS3/63)
  - The redirecting station is origination-restricted (CS3/43), or the call is redirecting on the first leg of a switch-classified call (CS3/42).
  - Calls that result in intercept treatment will not be allowed to redirect even if normally such calls would be sent to an attendant (CS3/63).

Protocol Error (NAK) Causes

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

With this service, when a call is routed to a new number, if the new number is a station on the switch, it begins alerting (if available to receive the call) and the originally-alerted call is dropped. At this point, this alerting call is eligible to be redirected again. If the new (redirected-to) number is off-premises, the call is routed there and the caller hears call progress tones. In this case, the call may not be redirected again.

For both on- and off-switch redirection, the switch will not drop the redirecting party until success is assured. It is at this point that the positive acknowledgement is sent. If the switch cannot redirect the call, NAK is provided and the alerting call will not be affected.

There is a special case where an adjunct can receive a positive acknowledgment, but the redirect may fail. For example, if the call is redirected to an off-premises number and the network trunk is seized, the switch considers this successful redirection and drops the redirecting party. The caller may hear any network-provided call progress tones.

Party_ids may be re-used if the call is redirected more than once.

An application using this feature should consider the timing for redirecting a call, particularly when redirecting away from a display user. If a display station user gets an incoming call and the application redirects it immediately, the station user may not have enough time to read the display (the incoming call information) before it is cleared by the application.

Since it is not possible to redirect busy calls, if possible an application should check whether the call had resulted in a busy condition before attempting to redirect it. This may be done if the application receives the appropriate call progress events from the switch (such as ‘‘busy,’’ ‘‘reorder/denial,’’ or ‘‘connected’’).

The call state may change between the Alerting event and the service request for redirection. An application should therefore expect certain NAKs and handle them appropriately.

Third Party Merge

The adjunct requests that two calls (already existing at the domain-controlled station) be merged.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Validates the parameters, sends an acknowledgement to the adjunct, and merges the two calls, or

— Denies the request if the parameters are invalid

Third Party Merge Parameters

common_party_id [ignored] Identifies the common endpoint with regard to the merged call. This endpoint must have one call on hold and one active (talking) state call.

call_id [mandatory] The held call_id to be merged with the active call.

call_id2 [mandatory] The active call_id to be merged with the held call.

conf/trans_flag The conf/trans_flag must be PRESENT. If set to TRANSFER, the call transfers and the controlled station is dropped from the call. The transfer operation merges the two existing calls and drops the controlled station from the call.

An acknowledgement is sent to the controlling adjunct (with party/extension information) when the transfer function is successful.

If the conf/trans_flag is CONF, the call conferences. The controlled station is retained in the resultant call.

ACK (Positive Acknowledgement) Parameters

An acknowledgement is sent with a call identifier for the merged call (call_id), a list of up to six party identifiers for the parties on the call (party_id), and a list of up to six extensions of the parties on the call to the controlling adjunct when the conference is complete.

result_call_id	[mandatory] The call_id after the call is merged (provided by the switch)
party_id	[mandatory] The list of party_ids for the merged call provided by the switch
Connected number	[mandatory] The list of extensions for the parties on the call

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid number/domain (CS0/28)
  The controlling party has not been specified correctly.
- Message not compatible with call state (CS0/98)
  The common party is not in a valid state for the operation (merge) to take place. For example, the common party does not have one call active (talking) and one call in the held state.
- Invalid information element contents (CS0/100)
  The party_id value of the request is invalid.
- Mandatory information element missing (CS0/96)
- Resources not available (CS3/40)
  The host attempted to add a seventh party to a call with six parties active (bridging and held cases only).
- Reorder/Denial (CS3/42)
  Both calls are alerting, or both calls are being service observed, or an active call is in a vector processing stage.

Protocol Error (NAK) Causes

The switch issues the following causes for returning a protocol processing error.

- Invalid association (CS0/81)
  The association does not exist.
- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The Third Party Merge capability may also be used by the adjunct to transfer to the supervisor. A sample scenario is presented below. Note that this is not the only way in which to effect a transfer to the supervisor.

An ACD agent handling the transfer may request (via data keyboard) that the call be transferred to the supervisor. The adjunct:

1. Places the existing call on hold
2. Issues a Third Party Make Call capability (with the ''supervisor-assist'' option) to the supervisor's extension
3. When the supervisor is either alerted or answers, the adjunct may merge the supervisor call and the previously held call using the Third Party Merge capability. (This capability is requested on the same association with the held call).

Since this was a transfer request, the conf/trans_flag parameter is set to TRANSFER and the switch drops the ACD agent from the connection after the merge takes place.

If the ACD agent requested a supervisor conference, the same procedure would have taken place but the conf/trans_flag would have been set to CONFERENCE. The switch does not drop the ACD agent from the resulting connection, and a three-party call is created.

Third Party Selective Drop

The adjunct uses this controlling capability to drop a controlled extension from a call.

Information Flow

The adjunct expects a response to its request.

The switch either:

— Drops the party once it has validated the parameters, or

— Denies the request if it could not execute the disconnect operation or if the call_id specified does not exist (invalid parameter).

Third Party Selective Drop Parameters

call_id [mandatory] Identifies the call to be disconnected

uui_info [optional] If an ASAI adjunct provides UUI in a 3rd Party Selective Drop, then the switch stores that UUI with the call.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Mandatory information element missing (CS0/96)
  The call_id is missing from the request.
- Invalid association (CS0/81)
  The association does not exist.
- Message not compatible with call state (CS0/98)
  The call is not currently active or in a hold state and therefore cannot be dropped.
- Invalid information element contents (CS0/100)
  The call_id value is invalid (for example, it is alerting).
- Invalid Information Element contents (CS0/100)
  The UUI IE is longer than 32 bytes.

Protocol Error (NAK) Cause

The switch issues the following cause for a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The capability may not be used to drop a party in the alerting or held state from a two-party or a multi-party call.

On a Domain Control association, only the Domain-Controlled extension can be dropped, not other parties as with the Third Party Selective Drop capability of the Call Control capability group (see Chapter 4, ''Call Control Capability Group'').

If a call has User to User Information (UUI) that came from an ISDN DISCONNECT message or in a Third Party Drop Request stored with it, and a party drops from the call and a Drop Event Report is sent, then the UUI is included in the report. The UUI passed in this request is only reported with other Drop Event Reports for the same call.

Third Party Auto Dial

An adjunct uses this capability to set up a two-party call between the domain-controlled station and an internal or external destination. This capability can only be requested by an application having an active domain-control association.

Information Flow

The adjunct expects an acknowledgement to its request only if the *return_ack* has been set to yes.

Third Party Auto Dial Parameters

dest_addr — [mandatory] This parameter specifies a valid on-PBX extension (station extension, VDN, ACD split, hunt group, announcement extension), or off-PBX number. May optionally contain the ARS/AAR if not present in the dest_route_select.

dest_route_select — This parameter contains the Trunk Access Code (TAC)/ARS/AAR information for off-PBX numbers.

return_ack — This parameter enables the switch to acknowledge the request made by the adjunct. It can be set to ‘‘yes’’ for acknowledgement or ‘‘no’’ for no acknowledgement. The default is ‘‘no.’’

priority_calling — [optional] If present, a priority call is placed if the destination is a local extension. If the priority flag is specified for an off-PBX destination, the call is denied. The default is non-priority.

uui_info — [optional] If an ASAI adjunct provides UUI in a Third Party Auto Dial, then the switch stores that UUI with the call.

UUI from an Auto Dial Call will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, and in a Route Request, if one is made.

ACK (Positive Acknowledgement) Parameters

If the **return_ack** was present in the adjunct's request, the switch will return an ACK containing the following:

party_id	The party identifier of originator
call_id	The call identifier of the resulting call

If the **return_ack** is not present, no ACK is returned.

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Third Party Relinquish Control

The adjunct uses this terminating capability to end a Domain Control association. The switch continues to process all calls at the domain-controlled station or split or EAS skill normally. Calls present at the domain-controlled station or split or EAS skill are not affected by this capability.

Information Flow

The adjunct expects a response to its request.

The switch always ends a domain control association and acknowledges the request.

Issuing this capability terminates the Domain Control association. Thus, all event reporting on the association ends.

This capability may be issued any time.

Third Party Relinquish Control Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause

None for this capability

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking a Third Party Domain Control Request over a Call Control association is inconsistent.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

Ending one Domain Control association does not affect the other active associations that may be controlling an extension, or the state of any calls at that extension.

This capability does not disconnect the domain-controlled station or any other station from any call.

Third Party Send DTMF Signals

The adjunct uses this controlling capability to transmit a sequence of DTMF tones on behalf of a domain-controlled party on the call.

Information Flow

The adjunct expects a response to its request.

The switch either:

- Transmits the tones on the sender's talk path after it has validated the parameters, or
- Denies the request if it is unable to service it

Third Party Send DTMF Signals Parameters

call_id — [mandatory] Identifies the call at the specified domain (station) for which DTMF signals are to be sent.

rx_party_id — [optional] If present, this parameter is ignored. A list of 1 to 5 parties who will receive the DTMF signals. If this list is null, then all parties on the call receive the DTMF signals if otherwise eligible (that is, if they are connected via ports that support end-to-end signaling). Currently not supported (ignored) by DEFINITY.

char_seq — [mandatory] Identifies the DTMF character sequence to be generated (maximum 32).

tone_dur — [optional] If present, this parameter is ignored.

pause_dur — [optional] If present, this parameter is ignored.

ACK (Positive Acknowledgement) Parameters

An acknowledgement is sent when it is determined that the DTMF digits could be transmitted, not when done transmitting.

None for this capability

Denial (NAK) Cause

The switch issues the following reasons as the cause for denying the request and ending the association:

- Mandatory Information Element missing (CS0/96)
  The char_seq is missing from the request. A denial may be sent if another parameter is missing as well.
- Invalid number/domain (CS0/28)
  Party_id is out of range or is a party on whose behalf ASAI cannot send DTMF tones (that is, a party other than a station or a trunk). Or, call_id is out of range.
- Invalid association (CS0/81)
  The request is supported only over call control and domain control associations. An attempt has been made to request this service over another type of association.
- Invalid Information Element contents (CS0/100)
  A value of an IE is outside the range specified, or a character sequence with a length of 0 or invalid characters has been supplied.
- Service or Option Not Available (CS3/63)
  The provided call ID does not exist.
- Message Not Compatible with Call State (CS3/98)
  DTMF signals can be generated only for active calls on which no other signaling tones are present. Also, the call must not be in vector processing or in a non-active state. The switch has detected that local, audible signals, including (PBX) dial tone, busy tone, ringback tone, intercept tone, or Music-on-Hold/Delay are currently being received. Or, the switch capacity for simultaneous sending of DTMF digits is full.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated or the capability invoked is not consistent with this association. For example, invoking the Third Party Make Call capability on a Domain Control association is inconsistent.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

- DTMF signals may be sent to any extension type on the call.
- If a character sequence with a length greater than 32 is supplied, only the first 32 characters in the sequence will be accepted (the remaining characters will be ignored).

Third Party Domain Control Ended

This capability is used by the switch to inform an application that a domain-control association has been terminated because the domain was removed or changed to become an invalid domain by administration.

Information Flow

The switch does not expect a response.

Third Party Domain Control Ended

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Cause

None for this capability

6 Event Notification Capabilities

This chapter describes the Notification Capability Group. The capabilities available in this group allow the adjunct to request and cancel event reporting on new calls.

The following capabilities are available:

Event Notification Request	This capability enables the adjunct to direct the switch to send event reports for new calls entering an ACD split or VDN domain.
Event Notification Cancel	This capability enables the adjunct to direct the switch to cancel event reports for a specified domain.
Stop Call Notification	This capability enables the adjunct to direct the switch to stop sending event reports for a chosen call within the domain.
Event Notification Ended	This capability allows the switch to notify the adjunct when a monitored domain becomes invalid (via administration) for event notification.
Event Report	See Chapter 3, ‘‘Common Capabilities.’’
U-Abort (User Abort)	See Chapter 3, ‘‘Common Capabilities.’’

Event Notification Request

This capability enables the adjunct to request notification for calls entering domains of VDNs and ACD splits. The requested ACD splits must not be adjunct-controlled[1] or vector-controlled.[2]

DEFINITY G3V3 and later allows a maximum of three notification associations to get events for a call.

Information Flow

The adjunct expects a response to its request.

The switch checks for a valid domain value and either accepts or denies the request.

Event Notification Request Parameters

domain type	This parameter must be a VDN or ACD split.
domain value	This parameter must be a valid VDN or ACD split extension. The requested ACD splits must not be adjunct-controlled (via switch administration) or vector-controlled.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues the following reason as the cause for not executing the request:

- Requested facility (capability) not subscribed/provisioned (CS0/50)
  The user has not subscribed for the requested capability.

1. Adjunct-controlled splits are ACD splits with this property administered on the Hunt Group form. Such splits accept only adjunct-originated calls, not off-net, incoming calls. Thus, event notification can be provided, but only via Third Party Make Call associations. See the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-204, for more information.
2. Vector-controlled splits are hunt groups that can only be accessed through a VDN. If notification is desired, it can be requested for the VDN that controls the split. See the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-204, and the *DEFINITY Communications System Generic 3 Call Vectoring Guide*, 555-230-520, for more information.

- Service or option not implemented (CS3/63)
  The given domain is already being monitored by up to 3 ASAI applications (G3V3 and later).
- Mandatory information element missing (CS0/96)
  A required parameter is missing in the request (for example, domain type).
- Resources not available (CS3/40)
  The request cannot be executed because the system limit would be exceeded for the maximum number of event notifications.
- Invalid Number/Domain (CS0/28)
  The domain value is neither a valid VDN nor a valid ACD split extension. The call is not monitored by the requesting association.
- Switch not administered correctly (CS3/41)
  The domain value is an adjunct-controlled split or a vector-controlled split.
- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of traffic overload. The adjunct or user may wish to retry the request but should not do so immediately.
- Invalid association (CS0/81)
  The association is already in existence.
- Service or option not available (CS3/63)
  A maximum of three notification associations to monitor an ACD split or VDN domain can be active for a specific ACD split or VDN.

Protocol Error (NAK) Cause

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

When monitoring is established for a domain, the switch generates event reports as calls arrive at the domain. The switch does not generate event reports for calls that are already present (in progress) at the domain when the Request Notification is received.

An ACD split or a VDN have can have up to three Event Notification Request associations active at any one time (G3V3 and later). Prior to G3V3, there could be only one.

A call can be reported on by up to three Event Notification associations at a time (G3V3 and later).

Event Notification Cancel

This capability enables the adjunct to cancel any notification request (for all calls) for a given domain only for the association that received the request.

Information Flow

The switch always accepts the request. If the request is not understood by the switch or is received over a non-Event Notification association, the switch aborts the association. See Chapter 3, ''Common Capabilities,'' for more information regarding the U-Abort capability.

Event Notification Cancel Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause

The switch issues the following reason as the cause for not executing the request.

- Invalid association (CS0/81)
  The association does not exist.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

The cancel notification takes effect immediately. Event reports to the adjunct for calls in progress cease over this particular association.

Stop Call Notification

This capability enables the adjunct to request that the switch stop sending event reports for a particular call, identified by the call_id parameter, over an Event Notification association.

Information Flow

The switch accepts or denies the request.

Stop Call Notification Parameter

call_id	This parameter indicates the call for which no further event notification is requested.

ACK (Positive Acknowledgement) Parameter

None for this capability

Denial (NAK) Causes

The switch issues the following reason as the cause:

- Invalid information element contents (CS0/100)
  The call_id parameter is outside the allowable range for the switch.
- Invalid number domain (CS0/28)
  The specified call is not monitored by the requesting association.

Protocol Error (NAK) Cause

The switch issues the following reason as the cause for not executing the requested operation:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

NOTE:
For more information regarding protocol errors, see the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

Only one call_id per request is allowed.

This capability can be requested by an adjunct at any time during the life of a monitored call. The request must be sent over the same Event Notification association that receives event reports for the call.

After this capability is invoked, a call may subsequently become monitored again if it enters another event notification domain or through other active event notification associations.

Event Notification Ended

This capability is used by the switch to notify the adjunct that, through switch administration, a domain with monitoring has become an invalid domain.

Information Flow

The adjunct does not respond to this capability.

The switch does not expect any acknowledgements (either positive or negative) from the informed adjunct and stops Event Notification for the calls automatically.

Event Notification Ended Parameter

Cause	Invalid Number/Domain (CS0/28) The domain is no longer available or valid.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause

None for this capability

Protocol Error (NAK) Cause

None for this capability

Considerations

The monitoring must have previously been requested by use of the Event Notification Request capability.

ASAI and Call Routing

7

This chapter describes the Routing Capability Group. The capabilities in this group allow the switch to ask for and receive routing instructions for a call. These instructions, issued by the adjunct, are based upon the incoming call information provided by the switch.

The following capabilities are available:

Route	This capability lets the switch ask the adjunct for the best route for an incoming call.
Route Select	This capability lets the adjunct answer the switch and provide route information for a call.
Route End	This capability lets the switch end the Route Request and informs the adjunct about the outcome of the route.
U-Abort (User Abort)	See Chapter 3, ‘‘Common Capabilities.’’

Route

This capability allows the switch to request routing information from the adjunct. The adjunct provides it based upon incoming call information. This feature may be used independently of or in conjunction with call monitoring (Event Notification turned on).

For G3V3 and later, the ASAI-Requested Digit Collection feature gives an adjunct the ability to request that a DTMF tone detector (TN744) be connected to detect user-entered digits. The request is made via an option of the Route Select message. The digits collected as a result of this feature are passed to ASAI monitoring and/or controlling adjuncts for action. The switch handles these digits like dial-ahead digits. The digits are collected while the call is not in vector processing; they are sent to an ASAI adjunct, and/or they may be used by Call Prompting features.

Information Flow

The switch sends a route request to the adjunct. The adjunct does not return an acknowledgment to the switch upon receipt of the routing request, but rather the adjunct sends a Route Select capability when a route is available.

Route Parameters

Calling Party Number/Billing Number (CPN/BN)

- For incoming call over PRI facilities — ‘‘calling number’’ from the ISDN SETUP message
- For incoming calls over non-PRI facilities, the calling party number is generally *not* provided. In this case, the Trunk Group number is provided instead.
- For calls originated at a bridged call appearance — the principal’s extension
- For incoming DCS calls, if the DCS calling party information is available to the switch (if a station with a display gets it), this information is also made available to ASAI. Otherwise, the calling party information is provided as the default.

Called Party number (DNIS)

- For incoming calls over PRI facilities, the Called Party Number is from the ISDN SETUP message.
- For incoming calls over PRI facilities to a VDN that does lookahead interflow on calls, if the lookahead interflow attempt fails, the called number provided is the principal extension of the dialed number.

- For incoming calls over non-PRI facilities, the Called Party Number is the principal extension [may be a group (TEG, PCOL, hunt group, VDN) extension[1]].

call_id	[mandatory] This parameter is the internal call identifier unique in the switch.
domain IE	[optional] This parameter is the VDN extension through which the Route Request is made.
user-entered information/ collected digits	[optional] This parameter represents the digits that may have been entered through call prompting or the ASAI-collected digits feature (G3V3 and later).
lookahead interflow	[optional] This parameter ensures that calls do not interflow to a remote location that cannot accept calls.
uui_info	[optional] UUI information received in an ISDN setup, ASAI Third Party Make Call, ASAI Auto Dial, or ASAI Route Select Message on the ASAI link is stored by the switch with the call for the life of the call or until overwritten by a previous Route Select associated with the call.
Flexible Billing	Specifies that the billing rate can be changed for an incoming 900-type call. Present if the feature is allowed for the call and the Flexible Billing customer option is assigned to the switch.

ACK (Positive Acknowledgment) Parameters

None for this capability

Denial (NAK) Causes

The adjunct might deny (NAK) a route request with adjunct-specific causes. See the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520, for a description of the adjunct routing vector step.

1. If the switch is administered to modify the DNIS digits, then the true DNIS is not passed.

Protocol Error (NAK) Causes

The adjunct might deny (NAK) a route request, if the request is invalid, with adjunct-specific causes. See the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221, for more information.

Considerations

A routing request is only administrable through the Basic Call Vectoring feature. (See the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-204, and the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520, for more information.) The Route capability is initiated by the switch when it encounters the **adjunct routing** command in a call vector. This command specifies an ASAI link's extension (adjunct) through which the switch sends the Route capability.

Multiple adjunct routing commands are allowed in a call vector. In G3V3 and later, the Multiple Outstanding Route Requests feature allows 16 outstanding Route Requests per call. The Route Requests can be over the same ASAI links or different ones. The requests are all made from the same vector. They must be specified back-to-back, without intermediate steps (wait, announcement, goto, or stop). If the **adjunct routing** commands are not specified back-to-back, pre-G3V3 adjunct routing functionality applies (that is, previous outstanding Route Requests are cancelled when an adjunct routing vector step is executed).

The first Route Select response received by the switch is used as the route for the call, and all other Route Requests for the call are cancelled.

If the adjunct denies the request (for example, replies with a NAK), the switch continues vector processing.

Event Reports for calls are not affected by the adjunct Route Request.

Route Select

This capability allows the adjunct to provide the switch with the destination address to which the call should be routed. In addition, the adjunct can request the switch to route the call as a direct-agent call and/or a priority call. The first Route Select received cancels all other outstanding requests.

For G3V3 and later, the following features provide additional functionality:

- **ASAI-Provided Digits** allows an adjunct to include digits in a Route Select capability. These digits are treated as dial-ahead digits for the call, which are stored in a dial-ahead digit buffer. They can be collected one at a time or in groups using the **collect digits** vector command(s).
- **ASAI-Requested Digit Collection** gives an adjunct the ability to request that a DTMF tone detector (TN744) be connected to detect user-entered digits. The request is made via an option of the Route Select message. The digits collected as a result of this feature are passed to ASAI monitoring and/or controlling adjuncts for action. The switch handles these digits like dial-ahead digits.

  These digits are collected while the call is not in vector processing. They are sent to an ASAI adjunct, and/or they may be used by Call Prompting features.
- **User to User Information (UUI)** allows distributed CallVisor ASAI and ACD users to associate caller information with a call. This information may be a customer number, credit card number, alphanumeric digits, or a binary string. It is propagated with the call whether the call is transferred or routed to a destination on the local switch or to a destination on a remote switch. Up to 32 bytes are allowed.

  An ASAI adjunct can include the UUI for a call in a Route Select. If the call is routed to a remote switch over PRI trunks, the switch sends the UUI in the ISDN SETUP message used to establish the call. The local and remote switches include the UUI in the Call Offered to Domain and Alerting Event Reports and in any Route Requests sent by the switch for the call.

Information Flow

The adjunct sends the destination address to the switch.

The switch accepts and reroutes the call if vector processing is executing either a wait time or announcement steps. If the destination address is invalid, the switch returns a Route End with cause CS0/28 (Invalid number) and continues vector processing (cancelling any ‘‘wait’’ or ‘‘announcement’’ steps in progress).

Route Select Parameters

orig_addr NA (ignored)

dest_addr [mandatory] This parameter is the valid destination for the call. If it is an off-PBX number, it can contain the u/ARS/AAR information.

dest_route_select [optional] Contains the u/ARS/AAR information for off-PBX destinations.

split_param If the direct-agent call option is set to ‘‘yes,’’ then this parameter must be a valid split extension; the destination address must be logged into this split.

direct_agent_call This parameter represents a special type of ACD call that is directed to a specific ACD agent rather than to any available agent. It may be set to ‘‘yes’’ or ‘‘no.’’

priority_calling This parameter represents a special type of call that carries three-burst distinctive ringing and does not go to the covering point for coverage or send all calls. It may be set to ‘‘yes’’ or ‘‘no.’’

user_entered_code Includes the following:

- Type of user code = customer-database provided (cdp)
- Collect/collected indication = collected
- Timer = all 0’s (default — not used)
- User data = ASCII digits (0-9, *, #)

collect_digits_flag This parameter indicates that digits should be collected (via a TTR in DTMF mode).

party_id This parameter indicates which party on the call the tone detector should listen to for ASAI-Requested Digit Collection. Currently, the call ‘‘originator’’ is the only option supported. (If present, this parameter is ignored.)

specific_event This parameter indicates when the tone detector used by the ASAI-Requested Digit Collection feature should be released. Options are ‘‘far end answer/connect’’ and ‘‘party disconnect.’’ Only the ‘‘far end answer’’ option is currently supported (starting with Release G3V3); other values are rejected. When the event option is not present in the Route Select and the digit collection is specified, the default is ‘‘far end answer/connect.’’

number_of_digits This parameter indicates the total number of digits that will be collected (1 to 24). For ASAI-Requested Digit Collection, every ‘‘#’’ and ‘‘*’’ count as one digit each. If the request is not in the valid range, the Route Select fails with cause CS0/79 — Service/option not available.

digit_coll_timeout This parameter indicates how many seconds (1 to 31) the tone detector will continue to collect digits after the first digit is received for ASAI-Requested Digit Collection. The default is no timeout.

uui_info [optional] If an ASAI adjunct provides UUI in a Route Select, then the switch stores that UUI with the call. The UUI overwrites any previous UUI stored with the call.

UUI from a Route Select will be sent in any ISDN PRI setup for the call, in the Alerting and Call Offered Event Reports, or in a future Route Select.

ACK (Positive Acknowledgment) Parameters

None for this capability

Denial (NAK) Causes

The switch issues a route end with one of the following reasons as the cause for denying the request. Vector Processing continues at the next step.

- When Route Select has hunt group as the destination and the hunt group is in night service and the night service destination is busy, then the cause value CS0/16 (Normal clearing) is generated when in fact the destination is busy (CS0/17).
- Invalid Association (CS0/81)
  This is generated when a Route Select is received after the Route End.
- Timer expired (CS0/81)
  The switch does not accept digits from a Route Select received after the corresponding Route Request was cancelled.
- Dial-ahead digits in incorrectly built Route Select (CS0/96 or CS0/100)
  If dial-ahead digits are received in an invalid Route Select message, they are discarded and a Route End is sent to the ASAI adjunct.
- Route Select with no called number and no dial-ahead digits (CS0/96)
  A Route Select received without dial-ahead digits and without a called number or with an empty called number is denied.
- Invalid party_id (CS0/28)
  If the party_id parameter is not valid for the call, the switch sends a Route End and continues with vector processing for the call.

- Permission denied (CS3/43)
  Only incoming trunks (of any type, including ISDN, MFC, and R2MFC) are eligible for ASAI-Requested Digit Collection. If the originator is not a trunk, a Route Select with the ASAI-Requested Digit Collection option is denied.
- Permission denied (CS3/43)
  A call prompter/tone detector is not connected if the originating trunk (to which the tone detector is to be connected) does not have incoming disconnect supervision administered. The Route Select fails, the switch sends a Route End, and the call continues in vector processing.
- Call has been terminated (CS0/86)
  If the call drops (for example, the caller abandons, a vector disconnect timeout occurs, a non-queued call encounters a ‘‘stop’’ step, or an adjunct clears the call), all outstanding Route Requests are cancelled.
- Timer Expired (CS0/102)
  The switch does not accept digits from a Route Select received after the corresponding Route Request was cancelled.

Protocol Error (NAK) Causes

None for this capability

Considerations

If the Route Select is received after a Route End is sent, it is ignored by the switch and vector processing continues.

Route End

This capability is sent by the switch to terminate the routing association and inform the adjunct regarding the outcome of the route.

Information Flow

The switch does not expect a response to the Route End.

Route End Parameter(s)

cause The switch issues a Route End for one of the following reasons:

- User busy (CS0/17)
  The destination is busy and does not have coverage. For this case, the caller hears either a reorder or busy tone.
- Call with requested identity has been terminated (CS3/86)
  The call was dropped while waiting for a routing response.
- The call has been routed successfully (CS0/16)
- The call has been redirected (CS3/30)
  The switch has canceled/terminated any outstanding Route Requests for the call after receiving the first valid Route Select message. The switch sends a Route End with this cause to all other outstanding Routing associations for the call.
- Invalid Information Element Contents (CS0/100)
  Either the UUI information element was longer than 32 bytes, or the header was missing.

For the above causes, vector processing does not continue. In the following causes, vector processing continues.

- Cancelling Outstanding Route Requests
  All outstanding Route Requests for a call are cancelled when a step other than ‘‘wait,’’ ‘‘announcement,’’ ‘‘goto,’’ ‘‘stop,’’ or ‘‘adjunct route’’ is encountered (cause value CS0/102). In addition, blank vector steps do not cause outstanding Route Requests to be cancelled.

- Receiving a Valid Route Select (CS3/30)
  The switch cancels/terminates any outstanding Route Requests for the call after receiving the first valid Route Select message[2]. The switch sends a Route End with cause CS3/30 (Call redirected) to all other outstanding routing associations for the call.

- Call Drops With Outstanding Route Requests (CS3/86)
  If the call drops (for example, caller abandons, vector disconnect timeout occurs, or a non-queued call encounters a ‘‘stop’’ step, or adjunct cleared call), all outstanding Route Requests are cancelled (cause value CS0/86 — Call has been terminated).

- Route — NAKs and Aborts
  Route Requests may be individually rejected (Route Request — NAK) or aborted (ABORT) by the adjunct without effect on other outstanding Route Requests.

- Vector Disconnect Timer Expires (CS0/86)
  When the Vector Disconnect Timer times out, all outstanding Route Requests are cancelled. Route End(s) is sent with cause CS0/86 — Call disconnected.

- Invalid number/domain (CS0/28)
  The destination address in the Route Select is invalid.

- Permission denied (CS3/43)
  Lack of calling permission, for example, for an ARS call, insufficient Facility Restriction Level (FRL). For a direct-agent call, the originator’s COR or the destination agent’s COR does not allow direct-agent calling.

- Recovery on timer expiry (CS0/102)
  This occurs when vector processing encounters any steps other than ‘‘wait,’’ ‘‘announcement,’’ ‘‘goto vector,’’ ‘‘goto step,’’ or ‘‘stop’’ after the adjunct routing command has been issued, or if processing times out at the wait step. See the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-220, and the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520, for more information.

2. A valid Route Select is defined as a Route Select containing all the appropriate parameters (information elements). The contents of the information elements do not need to be correct.

- Agent not a member of split (CS3/11)
  Upon routing to an agent (for a direct-agent call), the agent is not a member of the specified split.
- Agent not logged in (CS3/15)
  Upon routing to an agent (for a direct-agent call), the agent is not logged in.

ACK (Positive Acknowledgment) Parameters

None for this capability

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Cause

None for this capability

Considerations

If a non-ISDN PRI call is routed successfully, the Called Party Number provided in the pertinent Event Report is the same number provided by the adjunct as the new destination of the call in the Route Select.

If the call is an ISDN-PRI call, then the Called Party Number provided in the Event Reports is the original called number provided in the ISDN setup message.

ASAI and Request Feature Capabilities

8

This chapter describes the Request Feature Capabilities Group. These capabilities allow the adjunct to request or cancel switch-controlled features.

The following capabilities are available:

Request Feature	This capability lets the adjunct invoke or cancel switch-controlled features.
U-Abort (User Abort)	See Chapter 3, ‘‘Common Capabilities.’’

Request Feature Capability Group

An adjunct uses this capability to request invocation of one of the following switch features:

- ACD Agent Features
  - Login
  - Logout
  - Work mode changes
- Call Forwarding
- Send All Calls (SAC)

Information Flow

The adjunct expects a response to its request.

The switch either acknowledges or denies the request.

Request Feature Parameters

feature_id Specifies the feature to be invoked:

- Agent Login (login agent to ACD split)
- Agent Logout (logout agent from ACD split)
- Change agent work mode (change work mode of ACD agent to another mode)
- Call Forwarding
- Send All Calls

feature_params Specifies parameters specific to each feature:

- ACD agent login
  - Login identifier (password)
  - ACD split extension
  - Agent extension
  - [optional] Work mode (corresponds to initial work mode; if not specified, defaults to Auxiliary work)

    After call work,
    Auto in, Manual in,
    Auxiliary work

- ACD agent logout
  - ACD split extension
  - Agent extension
- ACD agent change of work mode
  - ACD split extension
  - Agent extension
  - Work mode

    After call work,
    Auto in, Manual in,
    Auxiliary work
- Activate Call Forwarding
  - Forwarding extension
  - Forwarded to number
- Cancel Call Forwarding
  - Forwarding extension
- Activate Send All Calls
  - Extension (Activates SAC at this number)
- Cancel Send All Calls (SAC)
  - Extension (Deactivates SAC at this number)

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues the following reason as the cause for not invoking the requested feature:

- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of processor overload. The adjunct or user may wish to retry the request but should not do so immediately.
- Requested facility (capability) not subscribed/provisioned (CS0/50)
  The user has not subscribed for the specific capability.
- Mandatory information element (parameter) missing (CS0/96)
  A required parameter is not present in the request.

- Resources not available (CS3/40)
  The request cannot be executed due to a lack of available switch resources.
- Invalid number (CS0/28)
  An invalid ACD split or agent extension value has been designated in the request.
- Invalid association (CS0/81)
  The association is already in existence.
- Agent not member of split (CS3/11)
  The agent making the request is not a member of the specified split.
- Incorrect number of agent login digits (CS3/14)
- Agent not logged in (CS3/15)
  The agent is not logged in (applies only to agent login).
- Agent logged into another split (CS3/13)
  The agent is already logged into the maximum number of splits.
- In same state (CS3/16)
  The request puts the agent in the same state that he or she is currently in.
- User Busy (CS0/17)
  The agent is busy on another call. For the particular extension, the agent is active (talking) on a call when a login request is made — an agent cannot be logged in when in the active state.

  Also, if the station user goes off-hook, dials one digit to begin making a call, and then the adjunct sends a login feature request for that particular user, the switch denies the request.
- Service/Feature not available (CS3/63)
  The feature is not available for the extension entered as the split extension. Note that an invalid split extension can be a valid nonsplit extension (such as an agent extension) on the switch, but it is still denied.
- Temporary Failure (CS0/41)
  System failure
- Agent state inconsistent with request (CS3/12)
  A work mode change is requested for a non-ACD agent or the ACD agent station is maintenance busy or out of service.
- Split not Administered Correctly (CS3/41)
  A request has been denied by the switch to log in, log out, manual-in or change work mode to auxiliary work or after-call-work for a member of the auto-available split. Change work mode is accepted for a member of the auto-available split only when the mode is to change to auto-in.
- Feature Request Rejected (CS3/53)
  This value is generated whenever the switch cannot return queried feature information even though the feature information may be defined for the specified extension.

- Service or Option Not Implemented (CS3/79)
  This value is returned when the queried feature has not been defined for the specified extension.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error(s):

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference,* 555-230-221.

Considerations

For nonadjunct-controlled ACD splits, agent login, logout, or change of work mode may be done manually via the voice set or the adjunct. Agents active in adjunct-controlled splits must be logged in or logged out and change work modes via the controlling adjunct.

Login over ASAI is accepted only if the agent meets certain state conditions. These state conditions must be the same as if it is being done manually via a voice terminal. For example, if an agent is busy on a call, login is denied.

An agent receives a logout denial if he or she is the last agent logged into the split and there are calls currently in the queue, unless the split is vector-controlled. If the split is vector-controlled, the last agent can log out even with a call in queue.

Changes of work modes are accepted if they are allowed via a voice terminal.

For the agent login request, the switch reads only the first 16 bytes, but it does not restrict the input of additional bytes.

The following summarizes how ASAI work mode changes perform when an agent is busy (either on an active or held call, or being alerted):

auto-in to auto-in the request is ACKD
auto-in to manual-in the request is ACKD
auto-in to aux-work the request is NAKD (cause = User Busy CS0/17)
auto-in to after-call-work the request is NAKD (cause = User Busy CS0/17)

manual-in to manual-in the request is ACKD
manual-in to auto-in the request is ACKD
manual-in to aux-work the request is NAKD (cause = User Busy CS0/17)
manual-in to after-call-work request is NAKD (cause = User Busy CS0/17)

aux-work to aux-work the request is ACKD
aux-work to manual-in the request is NAKD (cause = User Busy CS0/17)
aux-work to auto-in the request is NAKD (cause = User Busy CS0/17)
aux-work to after-call-work the request is NAKD (cause = User Busy CS0/17)

after-call-work to after-call-work the request is ACKD
after-call-work to manual-in the request is NAKD (cause = User Busy CS0/17)
after-call-work to auto-in the request is NAKD (cause = User Busy CS0/17)
after-call-work to after-call-work the request is NAKD (cause = User Busy CS0/17)

When an agent does not have auto-answer configured, they are not considered busy when they are:

- off-hook and idle, or
- off-hook and in a dialing mode on a call appearance.

However, if auto-answer is configured, the agent is busy whenever one or more call appearances are not idle (in other words, they are busy when in a dialing state).

(Generic 3r Only)
Agents in auto-available splits receive a denial if the following request is made to:

- Log in
- Log out
- Manual-in
- Change work mode to auxiliary work or after-call-work for a member of the auto-available split

Change work mode is accepted for a member of the auto-available split only when the mode is to change to auto-in.

ASAI and Value Query Capabilities

9

This chapter describes the Value Query Capability Group. The capabilities available in this group allow the adjunct to request and receive information about the status or value of switch-controlled features and services.

The following capabilities are available:

Value Query	This capability lets the adjunct ask for information about switch resources.
Value Query Response	This capability lets the switch split an answer into multiple messages to the adjunct for long replies.
U-Abort (User Abort)	See Chapter 3, ‘‘Common Capabilities.’’

Value Query

The adjunct can use this capability to inquire about specific resources available on the switch.

Information Flow

The switch either:

- Issues the information as part of the acknowledgement to the adjunct
- Denies the request with the appropriate cause code
- Sends one or more value query response messages with a response, then issues an acknowledgement to terminate the association

Value Query Parameters

item [mandatory] Specifies the switch resource

Valid items are:

- ACD Agent Login Audit Query
- Calls Query
- Party_id Query
- MWI Status
- Extension Information Query
- Time of Day

item_params Contains additional item-specific parameters

ACD Agent Login Audit Query The ACD split extension

ACD Agent Status Query The ACD agent extension and ACD split extension

ACD Split Query The ACD split extension

Calls Query The Station extension

Call Classifier Query No additional parameters needed

Extension Query The extension number (any valid dial plan number)

Party_id Query The call identifier for an existing call

Station Feature Query	With the Station Feature query, the application passes an extension number and the feature to the switch. The extension need not be domain-controlled, but must be one supporting the particular feature. The following features may be queried: ■ Message Waiting Indication (MWI) ■ Send All Calls ■ Call Forwarding All positive responses reflect the on/off status of the feature as known to the switch.[1]
Station Status Query	The station extension
Time of Day	No additional parameters needed
Trunk Group Query	The Trunk Group access code

Internally Measured Data Value Query Parameters

data_domain	Indicates the internally measured data domain for which data is requested. The following values are supported: agent, split/skill, trunk group, and VDN.
domain_element	Indicates the specific element (e.g., agent) from the specified domain for which data is requested. For **split/skill** and **VDN data_domains**, this is always an extension number corresponding to the target element. For **agents**, this is either the logical agent login ID when EAS is optioned, the BCMS/VuStats Login ID when the BCMS/VuStats Login ID feature is optioned, or the extension of the agent's phone (in a traditional ACD environment).

1. Although in most cases a lamp is associated with the feature for visual status, the query provides the feature status inside the switch, not the lamp status. (The lamp status may reflect some other station's MWI or it may be displaying an agent's MWI, based on administration.)

In a non-EAS environment with BCMS/VuStats login IDs, the address field of a logical agent contains the login ID (in ASCII). This can be up to 9 ASCII characters, 0 to 9. If either a BCMS/VuStats login ID or EAS logical agent is used, then the domain type in this element should be login_id.

For **trunk groups**, this is the TAC of the target group.

split/skill_ref

[Optional] Indicates a specific split/skill (from 1 to 4) into which the agent is currently logged, from which data is being requested. For acd-calls, avg-acd-talk-time, shift-acd-calls, or shift-avg-acd-talk-time, the value ‘‘0’’ signifies that data should be summed or averaged across all splits the agent is logged into.

data_items

[mandatory] Indicates the data items within the selected data_domain for which data is being requested.

If an IMD query about an agent requests historic data on acd-calls or on average-acd-talk-time and specifies a split rather than requesting the data over all splits to which the agent belongs), then those items are not returned. Historic data on agents does not segment these items into data for each split.

In the lists that follow, one, two, or three letters may follow an item. An ‘‘S’’ indicates a ‘‘split reference’’ is used, an ‘‘A’’ signifies a split of ‘‘All’’ can be used, and an ‘‘H’’ indicates that historical data is available through interval collection.

Valid items are:

Agent Data Items:

acd-calls **H, S, A**	split-avg-after-call-time **S**
agent extension	split-avg-speed-of-ans. **H, S**
agent-name	split-avg-time-to-abandon **H, S**
agent state **S**	split-call-rate **H, S**
avg-acd-talk-time **H, S, A**	split-calls-abandoned **H, S**
avg-extension-time **H**	split-calls-flowed-in **H,S**
avg-acd-call-time **H, S**	split-calls-flowed-out **H, S**
call-rate **H**	split-calls-waiting **S**
extension-calls **H**	split-name **S**

Agent Data Items, cont'd.:

extension-incoming-calls	split-number **S**
extension-outgoing-calls	split-extension **S**
shift-acd-calls **S, A**	split-objective **S**
shift-avg-acd-talk-time **S, A**	split-oldest-call-waiting **S**
split-acceptable-svc-lvl **S**	split-%-in-service-level **H, S**
split-acd-calls **H, S**	split-total-acd-talk-time **H, S**
split-after-call-sessions **S**	split-total-after-call-time **H, S**
split-agents-available **S**	split-total-aux-time **S**
split-agents-in-after-call **S**	time-agent-entered-state **S**

Agent Data Items, cont'd.:

split-agents-in-aux **S**	total-acd-talk-time **H**
split-agents-in-other **S**	total-after-call-time **H**
split-agents-on-acd-calls **S**	total-aux-time **H**
split-agents-on-ext.-calls **S**	total-available-time **H**
split-agents-staffed **S**	total-hold-time **H**
split-avg-acd-talk-time **H, S**	total-staffed-time **H**
total-acd-call-time **H**	

Spit/Skill Data Items:

acceptable-service-level	call-rate **H**
acd-calls **H**	calls-abandoned **H**
after-call-sessions	calls-flowed-in **H**
agents-available	calls-flowed-out **H**
agents-in-after-call	calls-waiting
agents-in-aux-work	oldest-call-waiting
agents-in-other	%-in-service-level **H**
agents-on-acd-calls	split-name
agents-on-ext.-calls	split-extension
agents-staffed	split-number
avg-acd-talk-time **H**	split-objective
avg-after-call-time	total-after-call-time **H**
avg-speed-of-answer **H**	total-aux-time
avg-time-to-abandon **H**	total-acd-talk-time **H**

VDN Data Items:

acceptable-service-level	calls-offered **H**
acd-calls **H**	calls-waiting
avg-acd-talk-time **H**	non-acd-connected-calls **H**
avg-speed-of-answer **H**	oldest-call-waiting
avg-time-to-abandon **H**	%-in-service-level **H**
calls-abandoned **H**	total-acd-talk-time **H**
calls-flowed-out **H**	vdn-extension
calls-forced-busy-or-disc **H**	vdn-name

Trunk Group Data Items:	
avg-incoming-call-time **H**	outgoing-usage **H**
avg-outgoing-call-time **H**	%-all-trunks-busy **H**
incoming-abandoned-calls **H**	%-trunks-maint-busy **H**
incoming-calls **H**	trunk-group-name
incoming-usage **H**	trunk-group-#
number-of-trunks	trunks-in-use
outgoing-calls **H**	trunks-maint-busy
outgoing-completed-calls **H**	

interval Indicates the collection interval for which requested data should be reported. Intervals are:

- Current — Current interval only (indicated by omitting this parameter)
- Last — Last complete interval

Integrated Directory Database Data Value Query Parameters

data_domain [mandatory] Indicates the data domain for which data is requested. This is a value for *integrated directory*.

domain_element [mandatory] Indicates the specific extension [1-5] digits for which the corresponding administered name is requested.

ACK (Positive Acknowledgement) Parameters

The following ACK parameters are passed by the switch to the adjunct in response to the particular value query request:

ACD Agent Login Audit Query The switch responds with a sequence of value query response messages that contain the extension for each agent logged into the split. Due to the volume of information contained in this response, the ACK is the last piece of data provided to the adjunct. It indicates the conclusion of the response. For most other queries, the information is returned within the ACK.

ACD Agent Status Query The switch responds with work mode and idle/busy state of the ACD agent. A ‘‘busy’’ state is returned if the ACD agent is on any active call or if Call Forwarding/SAC is active for the station.

An ‘‘idle’’ state is returned if the ACD agent is not on any active call (includes off-hook idle).

ACD Split Query

The switch responds with the number of ACD agents available to receive calls through that split, the number of calls in queue, and the number of ACD agents logged in. The number of calls in queue does not include direct-agent calls.

Calls Query

The switch responds with the call_id(s) for the calls present at the primary extension. Information for a maximum of 10 calls present at the station is reported back to the adjunct. Additional calls are not reported. This query is valid for stations only.

The switch provides the following information in the value parameter of the acknowledgement to the Calls Query request:

call_id list is a list of call-identifiers for calls present at the principal’s station (includes calls present at bridged appearances on the principal’s station)

call_state list is a list of call states for each call present at the principal’s station. Each call can be in any of the following states:

- Dialing (initiate): A station on the call is off-hook originating a call or listing to dial tone.
- Alerting: The call is alerting (ringing). This also includes calls at simulated bridges of stations that are ringing.
- Connected (active): A call is active at the station (talking state). This includes active calls at a bridged or simulated bridged appearance.
- Held: A call that was put on hold.
- Bridged: A call is present at a bridged, simulated bridged, button TEG, or PCOL appearance, and the call is neither ringing nor connected (active) at the station.
- Other: All other call states, including conference pending, listening to tone, and so forth.

party_id list provides a party identifier for the station on each of the calls. Note that there is a one-to-one correspondence between the elements of this list and the above two lists (call_id and call_state list).

Call Classifier Query

The switch responds with the number of ''idle'' and ''in-use'' TN744 ports. The ''in-use'' number is a snapshot of the TN744 port usage.

Extension Information Query

The switch responds with the following information:

Extension_class specifies the extension number as:

1. Vector Directory Number (VDN)
2. Hunt group (ACD split)
3. Announcement
4. Data extension
5. Voice extension: The response further defines the endpoint assigned to the extension number into various station types:
   - Analog (includes off-premises station extensions)
   - Proprietary
   - BRI
6. ASAI
7. Logical Agent
8. Other (for example, modem pool)

If the extension provided in the query is an agent's login ID, the response is ''Logical Agent'' and the address field of the domain IE contains the physical station extension the agent is logged into.

Integrated Directory Database Query

The switch responds with the following information (if a name is administered for the queried extension number):

1. actual type of queried user
2. extension of queried device
3. name of queried device

Party_Id Query

Switch responds with the party_id and the extension number (on-PBX extensions — local parties) on the call.

The switch responds with the following information in the value parameter of the acknowledgement to the Party_id Query request:

party_id is a list of identifiers for the endpoints on the specific call (call_id).

extension_number is a list of extension numbers for each party on the call. A party on the call may be an on-PBX connected extension, an alerting extension, or a split hunt group extension (when the call is queued). When a call is queued on more than one split hunt group, only one split hunt group extension is provided in the response to such query. For calls that are alerting at various groups (for example, hunt group, TEG, and so forth), the group extension is reported to the adjunct. For calls that are connected to the member of a group, the group member's extension is reported back to the adjunct.

For off-PBX parties, the switch always provides an extension entry even though it may contain the default value. A default value of ##### is provided.

Station Feature Query

Status of MWI feature

A positive acknowledgement indicates status of the message waiting indicator expressed as ‘‘on’’ or ‘‘off,’’ and the status as known by the following switch applications:

- ASAI
- Property Management
- Message Center
- Voice Messaging
- Leave Word Calling

The MWI query is defined for the following:

- All types of stations
- TEGs
- Hunt groups

Status of Send All Calls feature

A positive acknowledgement indicates status of the send all calls feature expressed as ‘‘on’’ or ‘‘off.’’ The status is always reported as ‘‘off’’ when the extension does not have a coverage path.

The SAC query is defined for the following:

- All station types
- TEGs

Status of Call Forwarding feature

A positive acknowledgement indicates the status of the Call Forwarding feature expressed as ‘‘on’’ or ‘‘off’’ and the forwarded-to number.

The Call Forwarding query is defined for the following:

- All types of stations
- Hunt Groups
- TEGs

Station Status Query

The switch responds with an ‘‘idle’’ and/or ‘‘busy’’ state of the station. The ‘‘busy’’ state is returned if the station is active with a call. The ‘‘idle’’ state is returned if the station is not active on any call.

Time of Day

The switch response contains the year, month, day, hour, minute, and second.

Trunk Group Query

The switch responds with the number of ‘‘idle’’ trunks in the group and the number of ‘‘in-use’’ trunks. The sum of the ‘‘idle’’ and ‘‘in-use’’ trunks provides the number of trunks in service.

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for ending the association:

- Invalid association (CS0/81)
  The association is already in existence.
- Invalid number (CS0/28)
  An invalid parameter (split extension, trunk access code, agent extension, etc.) has been designated in the query.
- Requested facility not subscribed/provisioned (CS0/50)
  The user has not subscribed for the requested capability.
  The Internally Measured Data feature requires that ASAI and Internally Measured Data both be optioned.
- Call with requested identity has been terminated (CS3/86)
  The call for which the request was made is no longer active.
- Resources not available (CS3/40)
  The request cannot be executed. Sent in response to an Internally Measured Data query if the system limit on simultaneous IMD queries is exceeded.
- Agent not member of split (CS3/11)
  The agent is not a member of the specified split.

- Agent not logged in (CS3/15)
  The agent is not logged in to the specified split.
- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of traffic overload. The adjunct or user may wish to retry the request but should not do so immediately.
- Feature Request Rejected (CS3/53) — Station Feature Query
  This value is generated whenever the switch cannot return queried feature information even though the feature information may be defined for the specified extension.
- Service or Option not Implemented (CS3/79) — Station Feature Query
  This value is returned whenever the queried feature has not been defined for the specified extension.
- Facility Reject (CS0/29)
  The applications processor attempted an ASAI query with the billing change request feature specified in the Item IE. This is not allowed.
- Mandatory Information Element Missing (CS0/96)
  For an Internally Measured Data Value Query, either data_domain, domain_element, or a data_item is missing. For an Integrated Directory Database query, the domain_element is missing.
- Invalid Information Element Contents (CS0/100)
  For an Internally Measured Data Value Query, the split reference or interval is outside the ranges specified for each information element. For an Integrated Directory Database query, the domain element is not one of the supported types. Also sent on an Internally Measured Data query and an Integrated Database query if the switch release is earlier than G3V4.
- Incompatible options (CS3/80)
  For an Internally Measured Data Value Query, an object is specified in a domain element with a type not matching that in the data domain, or an object on the switch is not the specified type.
- Unassigned Number (CS0/1)
  An item in an Internally Measured Data Value Query does not exist.
- Service not Available (CS3/63)
  The agent, split/skill, trunk group, or VDN queried in an Internally Measured Data Value Query is not measured.

The following is a list of instances where a CS3/79 is returned for a specific feature.

A negative acknowledgement with cause value CS3/79 is returned whenever the MWI has not been defined for the specified station. Examples of extensions for which the MWI query is not defined are:

- Attendant extension
- Data extension
- VDN extension
- Announcement extension
- DCS extension

A negative acknowledgement with cause value CS3/79 is returned whenever the Send All Calls feature has not been defined for the specified extension. A partial list of extensions not supporting Send All Calls is:

- Hunt groups
- VDNs
- Announcements
- Data modules
- DCS extensions

A negative acknowledgement with cause value CS3/79 is returned whenever the Call Forwarding feature has not been defined for the specified extension.

A partial list of extensions not supporting Call Forwarding is:

- VDNs
- Announcements
- PCOLs
- Data extensions
- DCS extensions

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error:

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

NOTE:
For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

For the Internally Measured Data (IMD) and Integrated Directory Database (IDD) features, applications are responsible for controlling traffic on ASAI links. They should attempt to minimize traffic in the following ways:

- For the IMD feature, they should request only needed data items, poll as infrequently as possible, and retain data items in their own memory rather than retrieve static data items from the switch.
- For the IDD feature, they should request names only when needed. For example, the IDD feature is not intended for use by an application to create its own copy of the ID database (by effectively downloading the entire ID database). Note that returned names may not be complete since names are a maximum of 15 characters in the ID database.

Since the IMD feature does not provide a way of informing an application which ACD operating environment is in effect (EAS, IMD Login IDs active, or traditional, no login IDs), the application must permit itself to be administered to adapt to these possibilities. The main impact is on the information sent to identify a target agent. In a traditional environment this is an extension, but in the other two cases, listed previously, appropriate login IDs must be used in the Query message.

Concurrent operation of VuStats and the IMD feature reduces the operating capacities of both. This is a function of the load on the switch CPU.

For Internally Measured Data:

- Numerical items that result in a division by zero are returned, but the IE holding that item has a length of zero and no data bytes (for example, average acd talk time, when no ACD calls have been made).
- Numerical items are returned in base 128. The most significant bit is first, with an extension bit set in the last byte.
- ASCII items are returned with an extension bit set on the last byte. Thus, only 7-bit ASCII values are supported.
- Certain data items may not be available when a query for IMD is made. If so, these data items will not be returned. Other items in the same request will be unaffected. The association is terminated with the normal RELease COMPlete Acknowledge message. For example, percent-in-service-level will not be returned if the service level is not administered.
- If all of the requested items are unavailable, then the association is still terminated with the RELease COMPlete Acknowledge. In that case, no FACility messages are sent, since there are no data items to return.
- VuStats (G3V4 Enhanced) is required to receive those data items that are part of the G3V4 enhancement to VuStats. The affected items are: call-rate, split-call-rate, total-acd-call-time, and average-acd-call-time. If VuStats (G3V4 Enhanced) is not enabled on the DEFINITY Customer-Options form and one of these items is requested, then the affected items will not be returned, as in the above bullet item.

Value Query Response

This capability is used by the switch to provide the adjunct with multiple responses to requested switch service information.

Currently, the switch replies to an ACD Agent Login Audit Query with up to eight agent addresses per Value Query Response message. The switch also replies to an Internally Measured Data Query with up to 8 data items per Value Query Response message.

Information Flow

The switch does not expect a response from the adjunct after sending a value query response.

Value Query Response Parameters

value The list of extensions corresponding to logged-in agents (ACD Agent Login Audit Query).

data_item Identifies the type of requested data item and its value. Up to 8 can be included in the Internally Measured Data Query message.

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Causes

None for this capability

ASAI and Set Value Capabilities

10

This chapter describes the Set Value Capability Group. The parameters available to this group enable the adjunct to set the value of certain switch-controlled services at an endpoint.

The following capabilities are available:

Set Value	This capability lets the adjunct set predefined values for switch-controlled features.
U-Abort (User Abort)	See Chapter 3, ‘‘Common Capabilities.’’

Set Value

The adjunct uses this capability to set the value of the message waiting indicator (MWI) and to set the billing rate of a 900-type call.

Information Flow

The adjunct expects a response to its request.

The switch either acknowledges or denies the request.

Set Value Parameters

item [mandatory] Specifies the item to be set. MWI and Flexible Billing are the only options presently available.

item_params [mandatory for MWI] For MWI, the on-PBX station extension of the party (station) for which the MWI is to be set

call_id [mandatory for Flexible Billing] For Flexible Billing, this switch-assigned call identifier is used to associate event reports and to identify a call that the adjunct wants to control. The call identifier is unique within the switch.

billing_type [mandatory for Flexible Billing] For Flexible Billing, one of the following must be present:

- New rate: Per minute rate that starts when message is set
- Flat rate: Time-independent rate
- Premium charge: Flat charge in addition to existing rate
- Premium credit: Flat negative charge in addition to existing rate
- Free call: Self-explanatory

amount [mandatory for Flexible Billing)] Applies to first four values of billing_type (Flexible Billing); not allowed for free call. Rate of call in dollars and cents. Cannot be less than $0.00 or greater than $999.99.

value [mandatory for MWI] For MWI, either on or off

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

The switch issues one of the following reasons as the cause for not setting the MWI:

- Invalid association (CS0/81)
  The association is already in existence.
- Invalid number (CS0/28)
  An invalid parameter value has been designated.
- Service or option not available (CS3/63)
  Messaging is not enabled for the requested station or would exceed ASAI request limits.
- Mandatory information element missing (CS0/96)
  A required parameter is missing in the request.
- Switching Equipment Congestion (CS0/42)
  The switch is not accepting the request at this time because of processor overload. The adjunct or user may wish to retry the request but should not do so immediately.
- Facility Reject (CS0/29)
  The applications processor attempted a Flexible Billing rate change request (Set Value capability) for a call that does not include a Flexible Billing user.
- Resources Unavailable, Unspecified (CS0/47)
  The applications processor requested a billing rate change and the switch threshold of unconfirmed requests has been reached.
- Message not compatible with call state (CS0/98)
  The billing change request is rejected because the call has not yet been answered or has been disconnected.
- No user responding (CS0/18)
  A response to a billing request has not been received.

Protocol Error (NAK) Cause

The switch issues the following cause for generating a protocol processing error(s):

- Protocol error (CS0/111)
  The Q.932 protocol has been violated.

  NOTE:
  For more information regarding protocol errors and a complete list of reason codes (cause values), see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Considerations

MWI

System cold starts cause the switch to lose the message MWI status. Hot starts (PE interchange) and warm starts do not affect the MWI status.

To keep the MWI in sync with the other adjuncts, the ASAI adjunct must use the Set Value capability to update the MWI whenever the link between the switch and adjunct comes up from a cold start.

When an ASAI adjunct has turned on a station's MWI and the station user retrieves messages using the station display, then the station display shows the message ''You have adjunct messages.''

The MWI can only be turned on for a station extension. Starting with G3V4, the Message Waiting Lamp on a station may track that physical extension or any other physical extension. This depends on the administration of the station. In an EAS environment, this lamp may track an agent's messages, depending on the system features administration.

The MWI cannot be turned on for an EAS agent through ASAI (a logical extension may not be specified in the ASAI request for turning on the MWI). Thus, this means that an ASAI request to turn on a particular MWL at a physical station may or may not result in that MWL actually turning on.

Flexible Billing

1. When an ASAI adjunct makes a Flexible Billing request, the switch normally passes the request onto a PRI trunk on the call. The PBX does not keep a timer while waiting for a response. ''No user responding'' is sent to the ASAI adjunct only when: 1) the PRI trunk providing the Flexible Billing service is dropped from the call before the response is returned to the PBX, or 2) a second billing request arrives before the first has been replied to.

   It is recommended that the adjunct keep timers to determine when a

Flexible Billing request has not received an answer in a reasonable time. The ‘‘No user responding’’ cause value from the switch will make the adjunct aware of the problem, but it will be too late to try again.

2. If VDN Return Destination is assigned, the Flexible Billing feature may not be able to function as desired because the new connection may result in the need for a different billing option that cannot be overridden, or may result in different rates that would inappropriately be applied to the previous connection(s).

   Thus if a call with the Returned Destination specified has the Flexible Billing trunk/call flag set, the Returned Destination is ignored and the call is forced to disconnect.

3. Flexible Billing works in a conference call arrangement as long as the identity of the incoming call is recognizable and the ASAI host controlling the billing change requests reside on the local switch.

   If two incoming calls are conferenced together, each with a Flexible Billing user, the operation of the feature is unpredictable if the proper incoming trunk call cannot be identified.

4. Subsequent rate requests work as follows:

   - If original and subsequent requests are ‘‘flat charge’’ or if both are ‘‘new rate’’, the amount of change is overwritten. The time stamp at which the change is to be applied remains the same as when the original change was requested.
   - If the original request is ‘‘Premium Charge’’ or ‘‘Premium Credit,’’ the subsequent request can be either ‘‘Premium Charge’’ or ‘‘Premium Credit.’’ The amount of the request and the type of request replaces the original request. The two requests are not arithmetically combined. The time stamp at which the change is to be applied remains the same as when the original change was requested.

ASAI Maintenance Capabilities

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This chapter describes the Maintenance Capability Group. The capabilities available in this group are used to disable and enable switch-administered alarms for periodic link maintenance and to obtain information about the condition of the ASAI link.

The following capabilities are available:

Heartbeat	This capability lets the adjunct and switch request a sanity check on the ASAI software or the ASAI link.
Suspend alarms	This capability lets the adjunct disable switch-administered alarms on the ASAI link before planned down time or routine maintenance.
Resume alarms	This capability lets the adjunct enable switch-administered alarms on the ASAI link.
Restart	This capability clears out all data structures and resources associated with the ASAI link (for example, associations).

CAUTION:
All adjuncts must support the restart and heartbeat procedures or the ASAI link will not operate. Further, it is strongly recommended that all adjuncts incorporate suspend/resume maintenance to avoid unnecessary alarms and resulting maintenance expenses when adjuncts are brought down for routine maintenance or normal shutdown.

Heartbeat

This capability enables the adjunct or the switch to send an application-to-application message and receive a response in order to determine the sanity of the application on the remote endpoint.

Information Flow

The sender expects a response to its request.

The switch responds to the Heartbeat within 60 seconds.

The switch sends the Heartbeat message over each established signaling link to an adjunct every two minutes. The adjunct is required to respond to the switch within ten seconds. Failure to respond to three consecutive heartbeats results in the switch bringing the link down and attempting to bring it back up again.

Heartbeat Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Causes

None for this capability

Protocol Error (NAK) Causes

None for this capability

Suspend Alarms

This capability enables the adjunct to disable switch alarms on an ASAI link for maintenance functions.

NOTE:
Since unnecessary alarms can result in unnecessary maintenance expenses, it is recommended that all adjuncts request this capability during the shutdown sequence.

Information Flow

The adjunct makes the suspend alarms request for that particular link.

The switch issues an ACK to notify the adjunct that it will not raise any alarms for that ASAI link.

Suspend Alarms Parameters

None for this capability

ACK (Positive Acknowledgement) Parameters

None for this capability

Denial (NAK) Cause(s)

None for this capability

Protocol Error (NAK) Cause(s)

None for this capability

Considerations

The Suspend Alarms capability overrides any administered alarms and halts periodic switch maintenance for the particular ASAI interface over which it is received.

When alarms are suspended on a link, the switch continues to service that ASAI link as follows:

- The switch continues to send periodic Heartbeat requests on any link that has suspended alarms, but does not raise alarms or attempt to re-initialize the link if there is no response to a Heartbeat request.
- The switch continues to process ASAI associations that are in progress.
- The switch accepts and processes requests for new ASAI associations.

Transmitting a Suspend Alarms on an ASAI link does not affect any associations in progress on the link. This allows any remaining associations to terminate gracefully. New associations may be initiated when alarms are suspended.

Resume Alarms

This capability enables the adjunct to resume switch alarms on an ASAI link.

Information Flow

The adjunct makes the suspend alarms request.

The switch issues an ACK to notify the adjunct that it will raise alarms for the ASAI link.

Resume Alarms Parameters

None for this capability

ACK (Positive Acknowledgement) Parameter(s)

None for this capability

Denial (NAK) Cause(s)

None for this capability

Protocol Error (NAK) Cause(s)

None for this capability

Considerations

If the adjunct does not acknowledge a Heartbeat request after sending Resume Alarms, the switch restarts the link and generates an alarm, if alarms are administered.

Restart

Restart provides that either endpoint may use this feature to free and reinitialize all resources for an ASAI interface. It also insures that if one ASAI endpoint detects a layer 2 drop (and therefore clears all its CRVs [for example, association] for the interface), ASAI messaging cannot continue on that interface without the other endpoint clearing its CRVs also.

Both the switch and the adjunct initiate the Restart when:

- An ASAI layer 2 link has been re-established after a link failure.
- A switch or adjunct maintenance subsystem determines a need to restart the ASAI interface.

If the switch cannot complete the Restart within a reasonable time and gets another Restart from the other side, it ignores the latter Restart until it gets another Restart.

Link Versions

This section applies to both the ASAI ISDN BRI and Ethernet links. With ASAI Versions, the switch sends a RESTART message to the adjunct containing the ASAI version(s) offered on the switch (currently V1 or V2). The adjunct responds to the switch's RESTART message with a Restart Acknowledgement message that includes the ASAI version of choice for that adjunct. If the switch does not support that version, the switch sends a RESTART message with a list of the available versions. If the adjunct sends a REstart Acknowledgement message back with no version, the link is initialized with default version 1. If the adjunct sends a Restart Acknowledgement message insisting on a version not supported by the switch, the link is not initialized and an error is logged.

Adjuncts without the versions procedure are able to perform pre-G3V4 RESTART procedures unchanged (in other words, RESTart — RESTart ACK). In these cases, the ASAI link comes up with the default version.

For G3V3 and later, the Versions and link initialization and restart procedures are covered in the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

ASAI and Feature Interactions

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This chapter describes the interactions between the ASAI capabilities and specific switch features.

Call Control and Domain Control do not prohibit users from access to any enabled switch features. Controlled stations can access any enabled switch feature.

Administration without Hardware (AWOH)

A station administered without hardware may be used as a station domain. However, no event reports are provided to the adjunct for this domain since there is no activity at such an extension. This feature should not be used with ASAI except for limited testing.

Analog Sets

Redirection

Analog sets do not support temporary bridged appearances. In normal circumstances, when a call at a multifunction set is left on a simulated bridge appearance, the call moves away from the analog set. Thus, any domain-control associations for the analog set receive the Call Redirected Event Report.

Alerting Event Reports are not sent to SAC-activated analog sets receiving calls.

Redirection on No Answer

Calls redirected by this feature generate the following event reports when a call is redirected from a nonanswering station:

- Call Redirected Event Report:
  Provided over the Third Party Domain Control associations when the call is redirected from a nonanswering agent. This event is *not* provided if the call is requeued to the split or delivered to another agent in the split.
- Queued Event Report:
  Generated if the call queues after being redirected.
- Call Ended Event Report:
  If the call cannot requeue after the call has been redirected from the nonanswering agent, then the call continues to listen to ringback until the caller is dropped. In this case, a Call Ended Event Report is generated when the caller is dropped and the call disconnected.

Direct-agent calls always redirect to the agent's coverage path instead of requeuing to the servicing ACD split.

Auto-Answer Option

The auto-answer analog sets do not receive dial tone after a Third Party Drop or Third Party Clear Call capability.

Manual Answer Option

Manual answer analog sets receive dial tone after receiving a Third Party Drop or Third Party Clear Call capability.

Number of Calls at Analog Sets

A maximum of three calls (one soft-held, one hard-held, and one active) may be present at the same time. In addition, the set may have a call waiting call.

A request to have more than three calls present is denied. For example, if an analog set user has three calls present and another call waiting, the user cannot place the active call on hold or answer the call. The only operations allowed are to drop the active call or merge the soft-held and active waiting call.

Number of Held Calls

A maximum of two calls may be in a held state at the same time. A request to have a third call on hold is denied.

Switch-Hook Operation

When an analog set goes on-hook with one or two calls on hold, the user is audibly notified. This is not reported as an Alerting event. When the user goes off-hook and is reconnected to the alerting call, a Reconnected Event Report is generated.

Going on-hook with a soft-held call and an active call causes the two calls to be transferred from the user's set. It does not matter how the held call was placed on soft hold.

Switch-Hook Flash Operation

If a controlled-extension analog user flashes the switch hook to put a call on soft hold to start a new call:

1. The Hold Event Report is sent to all monitoring associations.
2. A Call Initiated Event Report is returned to all Domain Controlled associations when the user receives the dial tone.
3. A Reconnect Event Report is returned to all associations if the user returns to the held call. If the held call is conferenced or transferred, the Conferenced or Transferred event reports are sent to all associations.

An analog set supports Third Party Merge requests even if the ‘‘switch-hook flash’’ field on the administration form is set to ‘‘no.’’

User-Classified Calls

If a user-classified call is placed for an analog set user without a speakerphone (or a headset), the user must either be idle or off-hook with dial tone, or go off-hook within five seconds of the call setup request. Otherwise, the request is denied (NAKed).

Direct-Agent Calls

For queued direct-agent calls, if the called destination agent has an analog set and is on-hook, the agent is notified with a ring ping; if the destination agent has an analog set and is off-hook and active on a call, the agent is notified with a call waiting tone regardless of the ‘‘Call Waiting Indication’’ option for the set. If the agent has an analog set and is off-hook but not active on a call, the agent does not receive audible notification for the direct-agent call.

An analog set cannot switch between a soft-held call and an active call from the voice set. However, with ASAI, this is possible by placing the active call on hard hold and retrieving the soft-held call.

Announcements

An Automatic Call Distribution (ACD) split forced first or second announcements and vector announcements do not generate event reports for the adjunct. However, nonsplit announcements generate events that are sent to other parties on the call.

Extensions assigned to integrated announcements may not be domain-controlled. The Third Party Make Call, Third Party Auto Dial, or Route Select capabilities may specify integrated announcement extensions as destination endpoints.

Answer Supervision

The ‘‘answer supervision timeout’’ field determines how long the central office trunk board waits before sending the (simulated) ‘‘answer’’ message to the software. This is useful when answer supervision is not available on a trunk. This message is used to: 1) send call information to Station Message Detail Recording (SMDR), 2) trigger the bridging of a service observer onto an outgoing trunk call, and 3) send a Connected Event Report for outbound calls placed on non-ISDN trunks. This message is ignored if the trunk is expected to receive true answer supervision from the network (the switch uses the true answer supervision whenever available). Adjunct-monitored calls are treated like regular calls.

With respect to switch-classified calls, when the ‘‘answer supervision’’ field is set to ‘‘no,’’ the switch relies entirely on the call classifier to determine when the call was answered. When answer supervision on the trunk is set to ‘‘yes,’’ a switch-classified call is considered ‘‘answered’’ when the switch software receives the ‘‘answer’’ message from the trunk board. In reality, switch-classified calls may receive either an ‘‘answer’’ message from the trunk board or (if this never comes) an indication from the classifier that the far end answered. In this case, the switch acts on the first indication received but not on any subsequent indications.

ARS/AAR

The ARS/AAR features do not change; they are accessible by ASAI adjuncts through Third Party Make Call, Third Party (Domain) Auto Dial, and Route Select requests. However, it is recommended that in situations where multiple applications use ARS trunks, ARS Routing Plans be administered using partitioning in order to guarantee use of certain trunks to the ASAI adjunct. Each partition should be dedicated to a particular application. However, this is not enforced by the switch.

When ARS/AAR is used, if the adjunct wants to obtain trunk availability information, it must query the switch about all trunk groups in the ARS partition dedicated for that application. The adjunct may not use the ARS/AAR code in the query to obtain trunk availability information.

When using ARS/AAR, the switch does not tell the adjunct which particular trunk group was selected for a given call.

Care must be given to the proper administration of this feature, particularly the FRLs. If these are not properly assigned, calls may be denied despite trunk availability.

The switch does not attempt to validate the ARS/AAR code prior to placing the call.

ARS must be subscribed if outbound calls are made over ISDN-PRI facilities.

Attendants and Attendant Groups

Individual attendants may be parties on adjunct-monitored calls and are supported like regular station users (all events are reported).

The attendant group is not supported with Third Party Make Call. It may never be specified as the originator and in some cases cannot be the destination.

An attendant group may be a party on an adjunct-monitored call, but the Alerting, Connected, and Disconnect/Drop Event Reports do not apply.

An attendant group extension cannot be a station domain.

An individual attendant extension number cannot be a station domain, but it can be a destination for a call originated from a station domain (event reports are sent about the individual attendant that is the destination for the call).

See "Attendant Control of Trunk Group Access" that follows.

Attendant-Specific Button Operation

This section clarifies what events are sent when the attendant uses buttons specific to an attendant console.

- Hold button
  If an individual attendant presses the hold button and the call is monitored or controlled, the Hold Event Report is sent to the corresponding association.
- Call Appearance button
  If an individual attendant has a call on hold, and the call is controlled and/or monitored, then the Reconnected Event Report is sent on the corresponding associations.
- Start button
  If a call is present at an attendant and controlled and/or monitored, and the attendant presses the start button, then the call is put on hold and a Hold Event Report is sent on the corresponding associations.
- Cancel button
  If a call is on hold at the attendant and the attendant presses the start button, putting the previous call on hold, and either dials a number and then presses the cancel button, or presses the cancel button right away, the call that was originally put on hold is reconnected and a Reconnected Event Report is sent to the association monitoring and/or controlling the call.
- Release button
  If only one call is active and the attendant presses the release button, the call is dropped and the Disconnect/Drop Event Report is sent to the association monitoring and/or controlling the call. If two calls are active at the attendant and the attendant then presses the release button, the calls are transferred from the attendant and a Call Transferred Event Report is sent to the association monitoring and/or controlling the calls.
- Split button
  If two calls are active at the attendant and the attendant presses the split button, the calls are conferenced at the attendant and a Call Conferenced Event Report is sent to the associations controlling and/or monitoring the calls.

Attendant Auto-Manual Splitting

If an individual attendant receives a call with active domain-control associations, and then activates the Attendant Auto-Manual Splitting feature, a Hold Event Report is returned to the associations controlling the extension’s adjunct. The next event report sent depends on what button the attendant presses on the set (CANCEL = Reconnect, SPLIT = Conference, RELEASE = Transfer).

Attendant Call Waiting

Calls that provide event reports over domain-controlled associations and are extended by an attendant to a local, busy, single-line voice terminal generate the following event reports:

- Hold — When the incoming call is **split away** by the attendant
- Connect — When the attendant returns to the call

The following events are generated if the busy station does not accept the extended call and its returns.

- Alerting — When the call is returned to the attendant
- Connect — When the attendant returns to the call

Attendant Control of Trunk Group Access

Third Party Make Call capability with the alert_order option (switch-classified calls) and Route Select capability requests cannot be offered to a Trunk Group with active attendant control. In this case, the call is terminated and a negative acknowledgement (NAK) is sent to the adjunct.

Calls that provide event reports over domain-controlled associations can access any Trunk Group controlled by the attendant. The attendant is alerted and places the call to its destination.

User-classified calls and forwarded supervisor-assist or direct-agent calls may terminate on trunks controlled by the attendant. Active-notification calls are also allowed to terminate on such trunks. Trunks seized for switch-classified Third Party Make Calls must not have attendant control activated. If they do, such calls are denied (cause CS0/21). For adjunct-routed calls, if the route select capability attempts to route to such a trunk, the step fails (cause CS0/21) and is skipped.

AUDIX

Calls that cover to AUDIX do not maintain a simulated bridge appearance on the principal's station. (This is true unless DEFINITY AUDIX is operating without a DCIU link. In this case, the simulated bridge appearance is maintained.) The principal receives audible alerting followed by an interval of coverage response followed by the call dropping from the principal's set. When the principal receives alerting, the Alerting Event Report is sent. When the call is dropped from the principal's set because the call went to AUDIX coverage, the Call Redirected Event Report is sent.

Authorization Codes

The switch negatively acknowledges (NAKs) (Call Rejected CS0/21) any Third Party Make Call capability for a switch-classified call requiring authorization codes. For all other controlled or active-notification calls, the originator is prompted for authorization codes.

Automatic Call Distribution (ACD)

Agents in Adjunct-Controlled Splits

Adjunct-controlled splits are ACD splits administered to be controlled by a single adjunct. Agents logged into adjunct-controlled splits have their voice set locked and must use Call Control and Request Feature capabilities to access telephony and ACD support features. Adjunct-controlled splits may not receive any nonadjunct-monitored calls; these are given a busy tone if they try to terminate at such splits.

When the ASAI link is down, adjunct-controlled splits behave like nonadjunct-controlled splits. Agents logged into such splits when the link is down have their voice sets unlocked without being logged out. When the ASAI link is restored, adjunct-controlled splits return to being adjunct-controlled, and agents' voice terminals become locked again.

Adjunct-controlled splits may also be vector-controlled. Splits that are both adjunct-controlled and vector-controlled have all the properties of both. Where there is conflict, the more restrictive property applies. For example, an agent logged into such a split cannot log into any other split; such an agent has the voice set locked. Nonadjunct-monitored calls are not allowed to terminate at such split.

An Event Notification request is denied (cause CS3/63) if the domain is an adjunct-controlled split.

Agents in Multiple Splits

An agent cannot be logged into multiple splits if that agent is logged into an adjunct-controlled split.

When an agent is logged into multiple splits, all direct-agent calls destined for the agent are serviced before all nondirect-agent calls in all splits. When there is more than one split with direct-agent calls waiting for the same agent, then the direct-agent call with the longest queue waiting time is serviced first.

Agent Status Displays

The agent status lamps reflect the agent’s current work mode, whether it was updated via the telephone set or via call control requests.

Announcements

Announcements played while a monitored call is in a split queue or as a result of an announcement vector command create no event reports. Calls made directly to announcement extensions have the same event report sent to the adjunct as calls made to station extensions. In either case, no Queued Event Report is sent to the adjunct.

If a vector routes a call to an announcement extension via the **route to** vector command, or if the Route Select capability routes a call to an announcement extension, event reports equivalent to station extensions are sent to the adjunct (assuming the call is an adjunct-monitored call).

If a user is hearing an announcement via an announcement vector command and a valid Route Select is returned, the announcement is stopped, but because this is an announcement while in vector processing, there are no event reports sent to the adjunct.

Assist/Supervisor Assist

This feature can be accessed in the conventional way from the voice set if the set is not locked. In this case, the call is placed to the switch-administered split supervisor.

If the set is locked (under adjunct control), this feature may only be accessed via the adjunct. This feature may also be accessed via the adjunct for sets that are not locked. It is initiated when the adjunct requests a Third Party Make Call with the assist flag set. The adjunct provides the supervisor extension. See Chapter 3 for interactions with other fields in the Third Party Make Call Request.

Whenever the ‘‘supervisor-assist’’ option is set in a Third Party Make Call, this call is measured by CMS in the same manner as the ACD ‘‘assist’’ call (provided the agent is logged into that split).

Automatic Answering

For direct-agent calls, receiving agents with automatic answering receive the single zip tone when the call is delivered, as in the case with regular ACD calls.

Interflow

This occurs when a split redirects all calls to another split on another PBX by activating off-premises call forwarding. This can also be done by vectoring with the **route to numbers** command to a destination off the PBX.

When an adjunct-monitored call interflows, adjunct notification ceases except for the Trunk Seized (for a non-PRI trunk) and Trunk Disconnect/Drop Event Reports.

Switch-classified calls can be interflowed but no call classification is done since the original call was to a local extension. The originator of the switch-classified calls should not be interflowed, since the adjunct would no longer be able to control the call.

Intraflow

This occurs when a call is redirected from one split to another split within the same PBX by following the split's coverage path. When an adjunct-monitored call intraflows, it retains the adjunct-monitored call status. Switch-classified calls can also be intraflowed.

Also, this occurs when a call is directed from one split to another by: 1) following the split's coverage path, 2) Call Forwarding, or 3) Call Vectoring via Queue-to-Main steps or Check Backup Split steps.

Note that the direct-agent call does not intraflow since it follows the agent's coverage path rather than the split's.

Night Service

Third Party Make Calls to splits in night service go to night service. A switch-classified Third Party Make Call originated from a split with the Night Service feature active is delivered to the night service extension. Calls in queue when this feature is activated remain in queue. The Call Offered to Domain Event Report is sent when a call that is not an adjunct-monitored call enters an ACD split (not adjunct-controlled) with active notification and also has night service active.

Direct-agent calls are routed to the hunt group's night service extension if night service was activated for the specified split, even if the priority calling option is enabled. This is the same as regular ACD calls.

Queue Status Displays/Indications

Adjunct-monitored calls (except direct-agent) are included in all existing measurements affecting queue status display and buttons.

Direct-agent calls are not included in any of the existing measurements affecting queue status displays and buttons.

Service Observing

An adjunct-controlled or adjunct-monitored call can be service-observed provided that service observing is originated from a voice terminal and the service observing criteria is met. A Connected Event Report is generated every time service observing is activated for an adjunct-monitored call. A Disconnect/Drop Event Report is generated when the observer disconnects from the call.

For a switch-classified call, the observer is bridged on the connection when the call is given to the service observed party. Unlike the ACD operation, the observer receives the warning tone *after* the bridging is complete (provided the warning tone option is administered system-wide).

For a user-classified call, the observer is bridged on the connection when the destination answers. When the destination is a trunk with answer supervision (includes PRI), the observer is bridged on when actual far-end answer occurs. When the destination is a trunk without answer supervision, the observer is bridged on after trunk cut-through (timeout) event.

Applicable events are ‘‘connected’’ (when the observer is bridged on) with the observer’s extension, and ‘‘dropped’’ when the observer drops from the call. In addition, the observer may manipulate the call via Call Control requests to the same extent as via the station set.

Automatic Callback on Busy/No Answer

This feature cannot be activated by the adjunct over the ASAI interface.

This feature can be activated by a controlled station user. The callback appears as an incoming call to the controlled station association having the same call_id as the call which had been queued on busy. This call may not be redirected via ASAI Redirect Call. If Automatic Callback is activated after Redirect Call, it will still apply to the principal, not the currently alerting party.

Switch-classified calls and adjunct-routed calls are not allowed to queue on busy Trunk Groups or stations.

Auto-Available Split

Auto-available splits are designed to keep agents logged in and available at all times. Consequently, ASAI agent login, logout, and change work mode requests are disallowed for agents in auto-available splits.

The switch denies (NAKs with cause CS3/41 — Split not Administered Correctly) any request from the adjunct to:

A. Change the work mode of a member of an auto-available split
B. Log out any member of an auto-available split.

An auto-available split can be a notification domain and members of auto-available splits (agents) can be domain-controlled stations. Auto-available splits may be administered as adjunct-controlled splits; however, in such a configuration, adjunct requests to log in, log out, or change work mode for an agent are denied with cause value CS3/41.

BCMS Login IDs

In an Internally Measured Data Query for data collected on an agent, the agent ID (specified in a domain IE) must be a BCMS login ID. Also, the address type in that domain IE must have the new codepoint indicating a BCMS login ID (001, 0011). However, the domain IE that indicates the query is for Internally Measured Data (type has codepoint 001, 0100) still has the codepoint (000, 0011) that specifies the query is about an agent.

If an Internally Measured Data query requests an agent's extension, the BCMS login ID is the value returned.

Bridged Call Appearance

A Domain Control station can have a bridged appearance(s) of its primary extension number appear at other stations. For bridging, event reports are provided based on the **internal** state of bridging parties with respect to the call. A call to the primary extension number alerts both the principal and the bridged appearance. Two or more ‘‘alerting’’ events are triggered, one ‘‘alerting’’ for the principal, and one ‘‘alerting’’ for each of the bridged appearances.

Two or more ‘‘connected’’ events may be triggered, if both the primary extension number and the bridged appearance(s) pick up the call. When the principal or bridging user goes on-hook but the bridge itself does not drop from the call, no event report is sent but the state of that party changes from the connected state to the bridged state. When the principal or bridging user reconnects, another Connected Event Report is sent. A Drop Event Report is triggered for the principal and each bridged appearance when the entire bridge drops from the call.

Members that are not connected to the call while the call is connected to another bridge member are in the ‘‘bridged’’ state. When the only connected member of the bridge transitions to the held state, the state for all members of the bridge changes to the held state even if they were previously in the bridged state. There is no event sent to the bridged user association for this transition.

Both the principal and bridging users may be individually domain-controlled. Each receives appropriate events as applicable to the controlled station. However, event reporting for a member of the bridge in the held state is dependent on whether the transition was from the connected state or the bridged state.

Call Control requests work normally if invoked over the station domain, regardless of whether both the principal and bridging user(s) are on the connection. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the principal of the bridge has an analog set or the exclusion option is in effect from a station associated with the bridge.

A Third Party Auto Dial or Third Party Make Call call always originates at the primary extension number of a user having a bridged appearance. For a call to originate at the bridged call appearance of a primary extension, that user must be off-hook at that bridged appearance at the time the request is received.

The Party ID Query only reports those members of the bridge in the connected state. If the entire bridge is in the held or alerting state, then only the principal is represented in the reply.

The Redirect Call is allowed either from the primary or the bridging user as long as the call is alerting. If successfully redirected, the alerting call will be dropped from both the primary and bridging user sets.

Busy Verification of Terminals

A domain-controlled station may be busy-verified. A Connected Event Report is provided when the verifying user is bridged in on a connection in which there is a domain-controlled station.

A Third Party Selective Hold request is denied if requested for the verifying user’s station.

Call Coverage

If a call that goes to coverage is monitored on an active-notification association for an ACD split or a VDN domain, the association receives the Call Offered to Domain, Alerting, and Connected Event reports.

For an alternate answering position that is a domain-controlled station, the Alerting and Connected Event Reports are returned to its domain-control association.

The Call Redirected Event Report is sent to the principal's domain-control association when an analog principal's call goes to coverage. The Disconnect/Drop Event Report is sent for the coverage station's domain-control associations when the call which had been alerting at both the principal and the coverage is picked up at the principal.

Switch-classified calls placed to local destinations whose coverage criteria are met do not go to coverage, they remain at the called party. Switch-classified calls delivered to originators whose coverage criteria are met follow the originator's coverage path (provided they are not priority calls).

Direct-agent calls follow the agent's coverage path rather than the split's. This is different from regular ACD calls.

An adjunct-routed call is allowed to go to coverage as usual. If the coverage is busy, the user hears a busy tone and the adjunct is notified with the Route End capability (cause CS0/16 — Normal Clearing).

If a call covers and rings at a station with domain-control, then the Alerting Event will indicate Call Coverage in a Cause IE. The possible causes are: Send All Calls Cover All or Go to Cover (CS3/31), Principal Station is Busy (CS3/26), and Principal Station Not Answering (CS3/28).

When a call is redirected via Redirect Call, the coverage timer is restarted. Thus, if the destination does not answer, the call will go to the coverage point for the principal. The subsequent Redirection No Answer timer is also restarted when a call is redirected from a coverage point via Redirect Call.

Call Coverage Path Containing VDNs

When a call is redirected to a station/split coverage path and the coverage path is a VDN, the switch provides the following event reports for the call:

- Call Redirected Event Report (for the Third Party Domain (Station) Control association monitoring/controlling the station):
  A call redirected event can also be sent if the call provides events to an Event Notification association and the VDN in the coverage path has Event Notification active. The VDN with active notification receives a Call Offered to Domain Event Report. If the VDN in the coverage path is not monitored (Event Notification active), then no call redirected event is sent to the active Event Notification association providing event reports for the call.
- Call Offered to Domain Event Report:
  This report is only sent if the VDN in the call coverage path has an Event Notification association active (is being monitored). If this is not the case, the report is not sent.

All types of calls (user-classified, direct-agent, and switch-classified) are permitted to follow the VDN in the coverage path if the coverage criteria has been met. The call to an on-switch originator of a switch-classified call is never permitted to go to coverage.

All other event reports associated with calls in a VDN (for example, Queued and Alerting Event Reports) are provided to all monitoring and controlling associations active for the call.

Call Forwarding All Calls

Users at controlled stations can activate and deactivate the Call Forwarding All Calls feature from the voice terminal or via an ASAI adjunct. Activation and deactivation from the voice set and an ASAI adjunct may be intermixed.

A Third Party Make Call, Third Party Auto Dial, or Route Select to a station with the Call Forwarding All Calls feature active redirects to the ‘‘forwarded to’’ station. No Call Redirected Event Report is sent on a Domain Control association for the forwarding station, since the call does not alert the extension that has Call Forwarding activated. This is only if the call was placed directly to the ‘‘forwarded to’’ station.

A direct-agent call forwards if the destination split has call forwarding activated.

While Call Forwarding is active at a station, agent status value queries will show that the agent is busy.

When an agent activates call forwarding, existing direct-agent calls waiting in the queue for the agent are *not* forwarded. Only new direct-agent calls entering the split after the agent’s activation are forwarded. (Note that regular ACD calls never follow the agent’s call forwarding.)

Switch-classified calls placed to local destinations with Call Forwarding All Calls active do not forward, but remain at the called party. Switch-classified calls delivered to originators with Call Forwarding All Calls active are forwarded, even if the forwarding number is off-PBX.

If a monitored call is forwarded off-PBX over a non-PRI facility, the Trunk Seized Event Report is generated.

If a call forwards and rings at a station with domain-control, then the Alerting Event will indicate Call Forwarding in a Cause IE (cause=CS3/32).

Redirecting via Redirect Call to an endpoint with Call Forwarding activated will fail.

Call Park

A controlled station can activate Call Park.

A call may be parked manually at a station by use of the ‘‘call park’’ button (with or without the conference and/or transfer buttons), or by use of the feature access code and the conference and/or transfer buttons.

When a call is parked by using the ‘‘call park’’ button without either the conference or the transfer buttons, there are no event reports generated. When the conference or transfer buttons are used to park a call, the Call Conferenced or Call Transferred Event Reports are generated. In this case, the ‘‘calling’’ and the ‘‘called’’ number in the Call Conferenced or Call Transferred Event Reports are the same — that of the station on which the call was parked.

When the call is unparked, a Connected Event Report is generated with the ‘‘calling’’ and ‘‘called’’ numbers indicating the station on which the call had been parked, and the ‘‘connected’’ number is that of the station unparking the call.

Call Pickup

A call alerting at a controlled station may be picked up using Call Pickup. The station picking up (either the principal or the pickup user or both) may be domain-controlled. A Connected Event Report is sent to all active associations on the call when this feature is used. When a pickup user picks up the principal’s call, the principal’s set (if multifunction) retains a simulated bridge appearance and is able to connect to the call at any time. No event report is sent for the principal unless the principal connects in the call.

When a call has been queued first and then picked up by a pickup user, it is possible for an adjunct to see a connected event without having seen any prior alerting events.

The switch does not allow call pickup to be used to pick up a switch-classified call that terminates on an internal station extension.

Call Vectoring

A VDN can be an active notification domain. It can also be the destination of a call placed by Auto Dial, or of a User-Classified call placed by Third Party Make Call. Interactions between event reporting and call vectoring are shown in the following table.

Table 12-1. Interactions Between Feedback and Call Vectoring

Vector Step or Command	Event Report	When Sent	Contents**
When Call Enters Vector	Call Offered to Domain*	encountered	**Note 1**
Queue to main	Queued or Reorder/Denial	successfully queues queue full	 cause - queue full
Check backup	Queued or Reorder/Denial	successfully queues queue full	 cause - queue full
Messaging split	Queued or Reorder/Denial	successfully queues queue full	 cause - queue full
Announcement	none		
Wait	none		
GoTo	none		
Stop	Call Ended***	encountered	cause
Busy	Reorder/Denial	encountered	cause - busy
Disconnect	Disconnect/Drop	facility dropped	cause - busy
Go To Vector	none		
Route To (internal)	Alerting	when successful	
Route To (external)	Cut Through/ trunk seized	PRI Interworking/ Non-PRI trunk seized	
Adjunct Routing	route****	encountered	**Note 1**
Collect digits	none		
Route To Digits (internal)	Alerting	when successful	
Route To Digits (external)	Cut Through	when trunk is seized	

Note 1: Included in the message are the called party, calling party (ANI/SID/CLI), prompting digits collected, lookahead interflow, optional UUI, optional trunk group number, and an indication that Flexible Billing is available.

* Only reported over an active notification association

** All event reports include an ASAI call_id.

*** Unless call is queued. If it is queued, no report is provided.

**** The ASAI *Route* capability and response are not event reports, but are initiated using the vector command ‘‘adjunct routing.’’

Table 12-1. Interactions Between Feedback and Call Vectoring *(continued)*

Vector Step or Command	Event Report	When Sent	Contents**
Converse Vector Command	Queued Event	if the call queues for the agent or automated attendant (VRU)	
	Alerting Event	when the call is delivered to an agent or the automated attendant	
	Connected Event	when the call is answered by the agent or automated attendant	
	Drop Event	when the call disconnects from the agent or automated attendant	

Note 1: Included in the message are the called party, calling party (ANI/SID/CLI), prompting digits collected, lookahead interflow, optional UUI, optional trunk group number, and an indication that Flexible Billing is available.

* Only reported over an active notification association

** All event reports include an ASAI call_id.

*** Unless call is queued. If it is queued, no report is provided.

**** The ASAI *Route* capability and response are not event reports, but are initiated using the vector command ‘‘adjunct routing.’’

A call may not be redirected via Redirect Call while in vector processing.

Call Prompting

Up to 16 digits collected from the last **collect digit** vector command are passed to the adjunct in the Call Offered to Domain Event Report and the Route ASAI capabilities.

Lookahead Interflow

This feature is activated by encountering a **route to** vector command, with the route to destination being an off-PBX number, and having the ISDN-PRI, Vectoring (Basic), and Lookahead Interflow options enabled on the Customer Options form.

For the originating PBX, the interactions are the same as for any call being routed to an off-PBX destination by the **route to** vector command.

For the receiving PBX, the lookahead interflow Information Element in the ISDN message is included in all subsequent Call Offered to Domain Event Reports and Route ASAI capabilities for the call, when the information exists, and when the call is adjunct-monitored.

Adjunct Routing

Adjunct routing is only administrable by the call vectoring feature. When vector processing encounters an **adjunct routing** command in a call vector, a Route capability is invoked by the switch to request a preferred route. The adjunct provides a preferred route to be used by invoking the Route Select capability.

If the dest_addr in the Route Select provided by the adjunct is a valid extension and the call is a monitored non-ISDN PRI call, then the Called Party number provided in the Event Reports is that of the dest_addr provided by the adjunct in the Route Select. If the call is an ISDN PRI call, then the Called Party number provided in the Event Reports is the original called number provided in the ISDN Setup Message. This functionality is consistent with that of the route-to-digits-with-coverage vector step. For more information on this step, see the *AT&T DEFINITY Communications System Generic 3 Feature Description*, 555-230-204.

If the dest_addr in the Route Select capability is invalid or if the switch is unable to route the call to the dest_addr, a Route End along with the cause is returned to the adjunct. If vector processing encounters steps that queue the call, or if the call leaves vector processing, the switch sends a Route End to the adjunct indicating the termination of the request.

For G3V3 and later, the Multiple Outstanding Route Requests feature allows multiple route requests for the same call to be active at the same time. The Route Requests can be over the same or different ASAI links. The requests are all made from the same vector. They must be specified back-to-back, without intermediate steps (wait, announcement, goto, or stop). If the adjunct routing commands are not specified back-to-back, G3V2 routing functionality applies (that is, previously outstanding route requests are cancelled when an adjunct routing vector step is executed).

This capability increases the redundancy options available with ASAI. Previously, adjunct routing applications that wanted to have a backup link had to test whether or not the primary link was down and then execute the **adjunct routing** command for the backup link with a vector. With this enhancement, multiple adjuncts can route the call without waiting for the first route attempt to fail.

In addition, the application can use this feature to distribute the incoming call load evenly across adjuncts, based on the adjunct's CPU load.

Multiple Split Queuing

A Queued Event Report is sent for each split that the call queues to; therefore, multiple call queued events could be sent to the adjunct for one call.

If a call is in multiple queues and abandons (caller drops), one drop event (cause normal) is returned to the adjunct followed by a Call Ended Event Report or a Third Party Call Ended capability.

When the call is answered at a split, the call is removed from the other split's queue. No other event reports for the queues are provided in addition to the Alerting and Connected Event Reports.

Third Party Make Call

Third Party Make Call cannot have a VDN as the originator.

With the OCM/EAS feature enabled, a VDN extension can be an originator for a switch-classified call.

Vector-Controlled Splits

A vector-controlled split may not be used as a domain for *Request Notification*.

Call Waiting

When an analog station is administered with this feature and a call comes in while the user is busy on another call, the Alerting Event Report is sent to the adjunct. This call is eligible for redirection via Redirect Call.

Class of Restriction (COR)

Direct Agent Calling is a field on the Class of Restriction (COR) form that must be set to ''y'' for both the destination and originator. If either the originating or destination party of a direct-agent call does not have the proper COR, then the call is permitted but is not a direct ACD agent-type call.

In the case of adjunct routing, the COR of the associated VDN is used for calling party restriction checks.

For direct-agent calls, the agent's COR is used for the termination party restriction checks, whereas regular ACD calls use the split's COR for the termination party restriction checks.

Third Party Auto Dial and Third Party Make Call calls are originated by using the originator's COR.

For switch-classified calls, the COR associated with the ASAI link is used to determine permissions. If the COR check fails, the call is dropped, and a Call Ended Event Report is sent to the adjunct with cause CS0/21.

Class of Service (COS)

The Class of Service (COS) for the originator is never checked in conjunction with any ASAI capabilities.

Conference

Manual conference from a domain-controlled station is allowed, subject to the feature's restrictions. The Hold Event Report is provided as a result of the first button push or first switch-hook flash. The Conference Event Report is provided as a result of the second button push or second switch-hook flash, and only if the conference is successfully completed. On a manual conference, the Call Conferenced Event Report is sent to all the active associations for the resultant call. For conference from another association, the requesting association receives a positive acknowledgement (ACK), and all other associations for the call or endpoints receive the Call Conferenced Event Report.

See Chapter 3, ''Common Capabilities'' for more information on merged calls. For a complete list of cause values, see the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

Consult

When the covering user presses the Conference or Transfer feature button and receives dial tone, a Hold Event Report is returned to all adjuncts monitoring the call. A Call Initiated Event Report is then sent to the covering user on the domain-control associations monitoring the covering user. After the Consult button is pressed by the covering user, Alerting and Connected Event Reports are returned to associations monitoring the principal and covering user. The covering user can then conference or transfer the call.

Data Calls

Data calls cannot be originated via the Third Party Make Call or Third Party Auto Dial capabilities. Analog ports equipped with modems can be domain-controlled and calls to and from these ports can be controlled and monitored. However, Call Control capabilities may cause the call to be dropped by the modem.

DCS

With respect to ASAI event reporting, calls made over a DCS network are treated as off-PBX calls and only the Call Initiated, Trunk Seized, Call Ended, and/or Disconnect/Drop Event Reports are returned. DCS/UDP extensions that are local to the PBX are treated as on-PBX stations. DCS/UDP extensions connected to the remote nodes are treated as off-PBX numbers.

ASAI does not currently support DCS calls completely. DCS calls may or may not provide accurate information to an ASAI adjunct. In a pure DCS environment, if the DCS calling party information is available to the switch (if a station with a display gets it), this information is also made available to ASAI. Otherwise, calling party information is provided as the default (*****).

With DCS on ISDN, the same rule (as above) applies. This means that even though there may be information in the ISDN message, it is not passed to ASAI. Only calling party information from the DCS message is passed to ASAI (DCS prevails over ISDN).

Since there can be other side-effects of using ASAI in a DCS environment, it is best to avoid using such setups.

SAC or CF features may not be activated over an ASAI link for an off-PBX DCS extension.

Direct Agent Calling

Direct-agent is a special type of ACD call that is directed to a specific ACD agent, rather than to any available agent in the split. It is invoked by specifying the direct_agent_call option in the Third Party Make Call or Route Select capabilities. This section covers the similarities and differences between the direct-agent call and a regular ACD call; these similarities and differences are independent of whether the direct-agent call is adjunct-monitored or nonadjunct-monitored. Therefore, the following sections on Direct Agent Calling apply to both types of direct-agent calls. Direct Agent Calling may be invoked.

Agent Work Modes with ACD

All ACD agent work modes operate the same for direct-agent calls as for regular ACD calls; that is:

- An agent can answer a direct-agent call destined for him/her by becoming available in the split the direct-agent call is associated with; that is, the agent must be in the manual-in or auto-in mode for the split.
- While on a direct-agent call, the agent becomes unavailable for all subsequent direct-agent or regular ACD calls. Multiple call handling can override this.
- After disconnecting from a direct-agent call from a split that the agent was in auto-in before, then for each split the agent is logged into, the agent returns to auto-in if he/she was in auto-in mode before, and into manual-in if he/she was in manual-in mode before.
- After disconnecting from a direct-agent call from a split that the agent was in manual-in before, then the agent enters after-call-work mode for this split, and is considered unavailable for all other splits.

Agent Work Modes with EAS

- An agent can answer a direct-agent call destined for him/her by becoming available by selecting the manual-in or auto-in work mode.
- While on a direct-agent call, the agent becomes unavailable for all subsequent direct-agent or regular ACD calls. Multiple call handling can override this.
- After disconnecting from a direct-agent call in auto-in mode, the agent becomes available for all skills logged into.
- After disconnecting from a direct-agent call in manual-in mode, the agent enters after-call-work mode and is not considered available for direct-agent or skill ACD calls.

Priority Queuing

There are five possible levels of priority in split queuing. In increasing order of priority, these are: low, medium, high, top, and direct. The different priority levels can be obtained as follows:

1. Low priority:
   - With vectoring, queue to main command with low as priority
   - With vectoring, check backup command with low as priority
2. Medium priority:
   - With vectoring, queue to main command with medium as priority
   - With vectoring, check backup command with medium as priority
   - Without vectoring, calls to hunt groups
   - Without vectoring, nondirect-agent calls to ACD splits
   - Without vectoring, intraflowed ACD calls without priority on intraflow
   - Without vectoring, interflowed ACD calls without priority queuing on the incoming trunk's COR
3. High priority:
   - With vectoring, queue to main command with high priority
   - With vectoring, check backup command with high priority
   - Without vectoring, intraflowed ACD calls with priority on intraflow
   - Without vectoring, interflowed ACD calls with priority queuing on the incoming trunk's COR
4. Top priority:
   - With vectoring, queue to main command with top as priority
   - With vectoring, check backup command with top as priority

5. Direct priority:
   - Direct-agent calls to ACD splits

Incoming lookahead interflowed calls queue with the priority specified on the receiving vector.

For more information regarding the call vectoring commands, see the *DEFINITY Communications System Generic 3 Call Vectoring/EAS Guide*, 555-230-520.

The Priority Queuing option has no effect on the queuing of direct-agent calls. Direct-agent calls have priority over all nondirect-agent calls and are inserted ahead of all nondirect-agent calls in the split queue but behind previously queued direct-agent calls.

Therefore, an available agent can service a nondirect-agent call only if there are no direct-agent calls waiting for that agent in all the splits that the agent is logged into and available to receive ACD calls in. When there is more than one split with direct-agent calls waiting for the same agent, then the direct-agent calls with the longest queue waiting time from the splits in which the agent is available to receive ACD calls are serviced first.

Note that each queued direct-agent call occupies a queue slot administered on the Hunt Group form for the specified split.

Indications of Direct-Agent Calls in Queue

When a direct-agent call joins the split queue because the destination agent is active on a call, in the after-call-work or auxiliary-work modes, then the destination agent is notified as follows:

- Ring ping — If the agent has a multifunction set
- Ring ping — If the agent has an analog set and is on-hook
- 3-burst call waiting tone — If the agent has an analog set and is off-hook and active on a call

The 3-burst call waiting tone is given regardless of whether or not the ‘‘Call Waiting Indication’’ option is enabled on the analog set. The ring ping or 3-burst call waiting tone is given only once for each direct-agent call when the call queues. If the agent has an analog set and is off-hook but is not active on a call, then the direct-agent call queues without a call waiting tone.

In addition, the active work mode button lamp associated with the direct-agent call’s specified split, if administered on the destination agent’s voice set, also flashes (fast flutter) to indicate a direct-agent call is waiting. Flashing starts when the first direct-agent call enters the split’s queue for this agent, and stops when no more direct-agent calls are in the split queue waiting for this agent (besides being answered, direct-agent calls could also have been abandoned or covered). For example, if an agent in the manual-in work mode has a direct-agent call in queue and is active on an ACD call, then the agent’s manual-in work mode button for that split is fluttering; when the agent goes on-hook on the active call (thus going into after-call-work), then fluttering ceases on the manual-in button and picks up on the after-call-work button for that split.

Note that if the destination agent is not logged into the specified split, then the direct-agent call would have been rejected as described earlier in Chapter 3 under ‘‘Third Party Make Call’’ and in Chapter 7 under ‘‘Route Select.’’

Number of Calls In Queue

Direct-agent calls are not included in the calculation of number of calls queued for the split for any purpose, including: queue status indications, call vectoring command’s conditional threshold checks, ASAI split query, ASAI Queued Event Report, and monitor traffic hunt groups.

Oldest Call in Queue

Direct-agent calls are not included in the calculation of the length of time that the oldest call has been queued for any purpose, including: queue status indications, call vectoring command’s conditional threshold checks, and monitor traffic hunt groups.

Hunt Group Measurements

Direct-agent calls are included in all hunt group measurements, including list performance hunt group and list measurement hunt group.

Delivering Direct-Agent Calls

Zip tone also applies to direct-agent calls as it does to regular ACD calls. If the destination agent has the automatic answer option enabled on the set, in the ‘‘auto-in’’ or ‘‘manual-in’’ work mode for the specified split, and is off-hook with no active call appearance, then the direct-agent call is delivered with a zip tone.

As with regular ACD calls, when using the manual answering option, the ringer audibly alerts when a direct-agent call terminates to the destination agent.

ASAI may provide UUI for an outgoing ISDN-PRI call (included in the setup message) via make call or autodial or via an ASAI route select.

ASAI may receive UUI from an incoming setup message in a Call Offered or Alerting Event Report. Similarly, UUI may be specified in a Third Party Selective Drop for inclusion in the ISDN disconnect message, and UUI in a received disconnect message will be included in a Drop Event Report if one is sent.

Priority Calling

When the priority calling option is specified on the direct-agent call (via Third Party Make Call or Route Select), then the direct-agent call is delivered as a priority call (with priority ringing if the available receiving agent is on-hook), and the direct-agent call does not go to coverage (as with regular ACD calls with priority calling).

Priority calling is also supported with off-hook Third Party Make Call requests (for example, if user goes off-hook first, and then issues a Third Party Make Call request with priority flag set, the call is placed as a priority call).

Direct-Agent Coverage

If the split associated with the direct-agent call has call forwarding or night service activated for the split, then the direct-agent calls are forwarded. If the priority calling option is requested, the direct-agent call forwards with priority ringing at the night service destination. This interaction is the same as with regular ACD calls.

When a new direct-agent call successfully enters a split, if the destination agent has call forwarding or send all calls activated, then the direct-agent calls are forwarded. Note that this interaction is different from regular ACD calls. With regular ACD calls, the calls never follow the agent's call forwarding or send all calls. If the priority calling option is requested, then the direct-agent call forwards with priority ringing at the call forwarded destination, but does not cover in the case of send all calls when the priority calling option is requested.

When an agent activates call forwarding or send all calls, existing direct-agent calls waiting in the queue for the agent do not forward to the call forwarding or send all calls destination. Only new direct-agent calls entering the split after the agent's activation are forwarded.

Direct-agent calls follow the destination agent's coverage path. Note that this interaction is different from regular ACD calls. With regular ACD calls, the calls follow the split's coverage path rather than the agent's. If the priority calling option was requested, the direct-agent call follows the standard priority call rules for coverage, meaning the call does not go to coverage. Calls (either regular ACD or direct-agent) in queue remain in queue until the caller abandons or an agent answers.

The above interactions are summarized in the tables that follow.

Once the direct-agent call leaves the specified split or destination agent, it is no longer considered direct-agent; however, the call does not lose its adjunct-monitored property.

Table 12-2. Coverage Interactions for ACD Calls without Priority Calling

Without Priority Calling	Regular ACD Call	Direct-Agent Call
Split Night Service activated	forwarded	forwarded
Split Call Forwarding activated	forwarded	forwarded
Split Coverage activated	forwarded	—
Agent Call Forwarding activated	—	forwarded
Agent Send All Calls activated	—	forwarded
Agent Coverage activated	—	forwarded

Table 12-3. Coverage Interactions for ACD Calls with Priority Calling

With Priority Calling	Regular ACD Call	Direct-Agent Call
Split Night Service activated	forwarded	forwarded
Split Call Forwarding activated	forwarded	forwarded
Split Coverage activated	not forwarded	—
Agent Call Forwarding activated	—	forwarded
Agent Send All Calls activated	—	not forwarded
Agent Coverage activated	—	not forwarded

Do Not Disturb

Do Not Disturb can be activated by an ACD agent. Activation of this feature for the agent blocks both personal calls and direct-agent calls from terminating at the agent's station. Regular ACD calls are still delivered to the ACD agent when this feature is activated. This is because personal calls and direct-agent calls use the agent's COR for termination restriction checks, whereas regular ACD calls use the split's COR for termination restriction checks.

Drop Button Operation

The operation of this button is not changed with ASAI.

When the ‘‘Drop’’ button is pushed by one party in a two-party call, the Disconnect/Drop Event Report is sent with the extension of the party that pushed the button. The originating party receives dial tone and the Call Initiated Event Report is reported on its domain-control associations.

When the ‘‘Drop’’ button is pushed by the controlling party in a conference, the Disconnect/Drop Event Report is sent with the extension of the party who was dropped off the call. This might be a station extension or a group extension. A group extension is provided in situations when the last added party to a conference was a group (for example, a TEG, split, or announcement) and the ‘‘Drop’’ button was used while the group extension was still alerting (or was busy). Since the controlling party does not receive dial tone (it is still connected to the conference), no Call Initiated Event Report is reported in this case.

Duplication

The ASAI link is not affected by processor interchanges.

Expansion Port Network (EPN)

The Expansion Interface (EI) board (TN570) makes it possible for ASAIs to terminate on an Expansion Port Network (EPN) as well as on the Primary Port Network (PPN).

It is recommended that any ASAIs critical to a customer’s business terminate on the PPN to enable the ASAI to remain operational in the event of a fiber link or EI failure. Further, resources that are used by a critical ASAI adjunct such as classifiers, trunks, announcements, and agent ports should also home on the PPN for the following reasons:

- To keep these resources in service in the event of a fiber link or EI failure
- To minimize the amount of cross carrier traffic that could degrade ASAI response time and system performance

Expert Agent Selection (EAS)

Skill Hunt Groups

Skill hunt groups have the same functions and limitations as vector-controlled hunt groups (splits). From the ASAI adjunct perspective, both skill hunt groups and vector-controlled hunt groups are subject to the same ASAI rules and are treated identically. For example, skill hunt groups, like vector-controlled hunt groups, cannot be monitored directly (Event Notification Request) by an ASAI adjunct. The VDN providing access to the vector(s) controlling the hunt group can be monitored instead if event reports for calls offered to the hunt group are desired.

Logical Agents

A logical agent’s station extension (for example, physical station from which an agent logs in) can be domain-controlled (Third Party Domain Control) and all Third Party Call Control capabilities can be invoked on behalf of the agent’s station extension. All event reports applicable to switch stations apply to the logical agent’s stations. Login IDs, however, are not valid domains for the Third

Party Domain Control Capability. A request to domain (station) control a login ID is denied by the switch with the cause value CS0/28 — Invalid Number.

User-Classified Calls

For user-classified calls (Third Party Make Call, Third Party Auto Dial, and Route Select requests without a direct-agent, supervisor_assist, or alert_dest_first option):

- The orig_addr calling number may contain a logical agent's login ID or a logical agent's physical station. If the orig_addr contains a logical agent's login ID and the logical agent is logged in, the call is originated from the agent's station extension associated with the agent's login ID.
  If the orig_addr contains a logical agent's login ID and the logical agent is not logged in, the call is denied with cause value CS3/15 — Agent not logged in. The orig_addr may not contain a skill hunt group extension or a VDN.
- The dest_addr called number may contain any number a local station user might dial, including a logical agent's login ID, a logical agent's physical extension, a skill hunt group extension and a VDN providing access to a skill hunt group. If the dest_addr contains a logical agent's login ID, the call is originated as if the call had been dialed from the originator's voice set to the requested login ID. If the origination and destination CORs permit, the call is treated as a direct-agent call; otherwise, the call is treated as a personal call to the requested agent.

Direct-Agent Calls

The following ASAI interactions apply to direct-agent calls. For example, direct-agent calls follow the agent's station coverage path (do not follow a logical agent's coverage path) and if the specified agent is not logged into the specified split, the request is denied.

The calling number (orig_addr) of a direct-agent call may contain a logical agent's login ID or an agent's physical station. If the calling number contains a logical agent's login ID and the logical agent is logged in, the direct-agent call is originated from the agent's station. If the calling number contains a logical agent's login ID and the logical agent is not logged in, the direct-agent call is denied with cause value CS3/15 — Agent not logged in. If the calling number does not contain a station extension or a login ID, the request is denied with cause value CS0/28 — Invalid Number.

Logical Direct-Agent Calls

Logical direct-agent calls follow the same event reporting rules as any other call manually originated from a voice station. In addition, a logical direct-agent call to a login ID that does not have a logged-in agent follows the coverage path administered for the login ID.

The calling number (orig_addr) of a logical direct-agent call may contain a logical agent's login ID or an agent's physical station. If the calling number (orig_addr) contains a logical agent's login ID and the logical agent is logged in, the direct-agent call is originated from the agent's station. If the calling number contains a logical agent's login ID and the logical agent is not logged in, the direct-agent call is denied with cause value CS3/15 — Agent not logged in. If the calling number does not contain a station extension or a login ID, the request is denied with cause value CS0/28 — Invalid number.

Supervisor-Assist Calls

The calling number and called number for supervisor-assist calls (Third Party Make Calls with the supervisor_assist option) consist of the following:

- The calling number (orig_addr) may contain a valid station extension or a logical agent login ID. If the calling number contains a station extension, there must be a logical agent currently logged in at that extension and the logical agent must have a skill corresponding to the skill hunt group extension in the split parameter. If the calling number contains a login ID, that login ID must be currently logged in and the logical agent must have a skill corresponding to the skill hunt group extension in the split parameter. If the requested station extension does not have a logical agent logged in or if the requested login ID is not logged in, the request is denied with cause value CS3/15 — Agent not logged in.
- The called number (dest_addr) may contain any station or attendant extension, including a logical agent's physical station and a logical agent's login ID. If the called number contains a logical agent's login ID and the agent is not logged in, the request is denied with CS3/15 — Agent not logged in. If the called number contains a logical agent's login ID and the agent is logged in, the login ID is converted to its associated station extension.

Supervisor-assist calls placed to login ID destinations are not direct-agent calls; that is, they are not tagged as direct-agent calls for CMS and they do not exhibit direct-agent call station indications and queueing behavior.

Switch-Classified Calls

Switch-classified calls (Third Party Make Call with dest_alert first and service_circuit options) may be originated from skill hunt group extensions (orig_addr). However, there are no forced announcements, coverage, call forwarding and intraflow/interflow for these calls because all skill hunt groups must be vector-controlled hunt groups. Calls to vector-controlled hunt groups access the above features via the controlling vectors.

The called number (dest_addr) may contain a logical agent's station extension. It cannot contain a logical agent's login ID, a skill hunt group extension, or a VDN. If the called number contains a logical agent's login ID, a skill hunt group extension or a VDN, the request is denied with cause value CS0/28 — Invalid Number Format.

Event Reports

Whenever logical agents are part of a monitored or controlled call (for example, a call providing event reports to an ASAI adjunct), the following additional rules apply to the event report contents (items):

- The calling number (calling party number IE) always contains the logical agent's physical station number (extension), even though a Third Party Make Call or Third Party Auto-Dial request might have contained a logical agent's login ID as the originating number (orig_addr).
- The connected and alerting numbers (Connected Party number IE) contain the logical agent's station extension and never contain the login ID. This is true regardless of whether the call was routed through a skill hunt group, whether the connected station has a logical agent currently logged in, or whether the call is an adjunct-initiated or voice terminal-initiated direct-agent call.
- The called or dialed number (Called Party number IE) contains the number that was dialed, regardless of the station connected to the call. For example, a call may be alerting an agent station, but the dialed number might have been a logical agent's login ID, a VDN, or another station.
- The Call Conferenced and Call Transferred Event Reports are an exception to this rule. In these events the called number contains the station extension of the transferred to or conferenced party when a local extension is involved. When an external extension is involved, the called number contains the default extension (#####). If the transferred to or conferenced party is a hunt group or login ID, and the call has not been delivered to a station, the called number contains the hunt group or login ID extension. If the call has been delivered to a station, the called number contains the station extension connected to the call.
- The domain item in the Alerting and Queued Event Report for logical direct-agent calls contains the agent's first Primary skill logged into. This is the skill hunt group that logical direct-agent calls queue to. Note that the skill hunt group is provided, even though an adjunct-initiated, logical direct-agent call request did not contain a skill hunt group.

Logins and Logouts for Logical Agents

The following rules apply to logical agents' logins and logouts via the Request Feature capability:

- The password (user code IE) contains the logical agent's login ID and password as follows: login ID terminated by the pound sign (#) and followed by the password, if necessary. If the agent's password is not included, the pound sign may be omitted.
- The split parameter is ignored by the switch, since the agent has a predefined (administered) set of skills. However, the split parameter must be present in the request, since it is a mandatory parameter.

When EAS is disabled, the split parameter (domain IE) contains the ACD split to log in or log out the agent.

- In a login request, the agent_id or agent_extension (domain IE) parameter must contain the agent's physical station; it may not contain a login ID. If a login request is received with a login ID as the agent_id the request is denied with cause value CS0/28 — Invalid Number Format. In addition, if the station extension in the agent_id already has a logged-in logical agent, the login request is denied with CS3/16 — Agent in Same State.
- In a logout request, the agent_id or agent_extension (domain IE) may contain either the agent's physical station or the agent's login ID.

Work Mode Changes for Logical Agents

Since logical agents are defined to have a single work mode, a work mode change Request Feature applies to all skill hunt groups that a logical agent is logged into. The following rules apply to the Request Feature request parameters:

- The split parameter is ignored by the switch, since it does not contain useful information. However, the split parameter must be present in the request, since it is a mandatory parameter.
- The agent_id parameter may contain a station extension or a logical agent's login ID. If the agent_id contains a valid station extension, the station extension must have a logical agent currently logged in. If the agent_id contains a logical agent's login ID, the login ID must be currently logged in. If there is no logical agent currently logged in at a requested station extension or if a requested login ID is not logged in, the Request Feature request is denied with cause value CS3/15 — Agent not logged in.
- The work_mode parameter specifies the logical agent's new work mode for all the skill hunt groups associated with the agent.

Activate/Cancel Send All Calls and Call Forwarding

Call Forwarding and Send All Calls activation and deactivation request features may not be requested for logical agent login IDs. Call Forwarding and Send All Calls Request Features containing a login ID as the redirecting number (forwarding number) are denied with cause value CS0/28 — invalid number.

Call Forwarding and Send All Calls may be requested on behalf of a logical agent's station extension.

Value Queries for Logical Agents

In ACD agent status, station status, or call query Value Query requests, the extension or agent extension parameter may contain either a logical agent's station extension or a login ID. For all cases, the returned information applies to the station extension being used by the agent. If the extension parameter contains a login ID for a logical agent that is not logged in, the request is denied with cause value CS3/15 — Agent not Logged In.

In an ACD agent status Value Query, if the extension parameter contains a station extension and there is not a logical agent logged at the station, the request is also denied with the cause CS3/15 — Agent not Logged In.

In an ACD agent status Value Query, the split parameter is ignored, since a logged-in logical agent has a single work mode for all the skill hunt groups. Note, however, that the split parameter is mandatory and must be present in the request.

In an extension Value Query, the extension parameter may be a login ID. If a login ID is provided, the switch responds with a new extension type of logical agent (new codepoint, 001, 0011 for the Domain IE). If the agent associated with the login ID is logged in, the address field of the domain IE contains the station extension being used by the logical agent.

In an Internally Measured Data Value Query, the split/skill_ref parameter is useful in cases where an agent is logged into more than one split/skill.

- If it is omitted, then those requested data items that require a split/skill reference are not returned.
- If it is supplied but does not refer to a skill the agent is logged into, then those requested data items that require a split/skill reference are not returned. For example, if the agent is associated with two skills, and the skill reference is three, then data items that require a skill reference are not returned.

In an Internally Measured Data Query for data collected on an agent, the agent ID (specified in a domain IE) must be the agent's login ID. Also, the address type in that domain IE must have the new codepoint indicating a login ID (001, 0011). However, the domain IE that indicates the query is for Internally Measured Data (type has codepoint 001, 0100) still has the codepoint (000, 0011) that specifies the query is about an agent.

If an Internally Measured Data query requests an agent's extension, the login ID is the value returned. If an agent is not logged in, then the request is denied (cause=CS3/63, Service or Option Not Available).

The response to an agent login audit Value Query for a skill hunt group contains the list of logical agents currently logged into the specified skill hunt group. That is, the agent_addr_list (domain IE) in the response contains the logical agent's station extension for all logged in agents currently having the skill associated with the requested skill hunt group.

Facility Restriction Levels (FRLs)

Third Party Make Calls are placed using the originator's COR, the station's COR, or the split's COR.

Forced Entry of Account Codes

Third Party Auto Dial or Third Party Make Call call attempts to Trunk Groups with the Forced Entry of Account Codes feature assigned are allowed. It is up to the originating station user to enter the account codes via the touch-tone pad. Account codes may not be provided via the ASAI. If the originator of such a call is logged into an adjunct-controlled split (and therefore has the voice set locked), such a user is unable to input the required codes and will eventually get denial treatment.

Hold

Manually holding a call (either by using the Hold, Conference, Transfer buttons, or switch-hook flash) results in the Hold Event Report being sent to all active associations for this call, including the held extension. A held party is considered connected on the call for the purpose of receiving events relevant to that call.

Hot Line

A Third Party Auto Dial or a Third Party Make Call request made on behalf of a controlled extension that has this feature administered is denied by the switch.

Integrated Services Digital Network (ISDN)

The Third Party Auto Dial calls follow Integrated Services Digital Network (ISDN) rules for the originator's name and number. The Call Initiated Event Report is not sent for *en-bloc* BRI sets.

Last Number Dialed

The called party address (dest_addr) provided in a Third Party Make Call or Third Party Auto Dial capability is the last number dialed for the calling party (orig_addr) until the next call origination from the calling party. Therefore, the user can use the ‘‘last number dialed’’ button to originate a call to the destination provided in the last Third Party Make Call or Third Party Auto Dial capability. This does not apply to switch-classified calls.

Leave Word Calling

When activated at the caller's extension, Leave Word Calling will attach itself to the principal's extension, even if the call was redirected via Redirect Call.

Lookahead Interflow

When a DEFINITY switch attempts to send a monitored call to another DEFINITY using the Lookahead Interflow capability, the second switch may reject the interflow attempt. If this happens, any ASAI message associated with this call that contains Called Party Number information sent by the first DEFINITY will contain the administered extension of the VDN that received the call, instead of the original dialed number as presented by the network. If the interflowed call is accepted by the other switch, however, the Called Party Number presented in the Alerting and Connected event reports, by the first switch, will be the default trunk extension.

Multiple Split Queuing

When a call is queued in multiple ACD splits, the party query provides, in addition to the originator, only one of the split extensions in the party list. When the call is de-queued, the Alerting Event Report provides the split extension of the alerting agent. No other events are provided for the splits from which the call was removed.

Music On Hold

Music on hold (if administered and available) is given to a party placed on hold from the other end either manually or via the adjunct.

Personal Central Office Line (PCOL)

Members of a Personal Central Office Line (PCOL) may be domain-controlled. PCOL behaves like bridging for the purpose of ASAI event reporting. When a call is placed to a PCOL group, the Alerting Event Report is provided to each member's domain-control associations. The called number information passed in the alerting message is the default station characters. When one of the members answers the incoming call, the Connected Event Report provides the extension of the station that answered the call. If another member connects to the call, another Connected Event Report is provided. When a member goes on-hook but the PCOL itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state. The Disconnect/Drop Event Report is not sent to each member's domain-control associations until the entire PCOL drops from the call (as opposed to an individual member going on-hook). Members that are not connected to the call while the call is connected to another PCOL member are in the bridged state.

When the only connected member of the PCOL transitions to the held state, the state for all members of the PCOL changes to the held state even if they were perviously in bridged state. No event report is sent to any domain-control association(s) for bridged users for this transition.

All members of the PCOL may be individually domain-controlled. Each receives appropriate events as applicable. Call Control requests are not recommended for PCOL endpoints. Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the PCOL.

A Third Party Auto Dial or Third Party Make Call originates at the primary extension number of a user. For a call to originate at the PCOL call appearance of a primary extension, that user must be off-hook on the PCOL call appearance at the time the request is received.

If a party_id query is requested while the PCOL is alerting or on hold, one party member is reported for the group with the extension number specified as the default extension.

If a call query is requested on an extension while the PCOL call is active, only one call appearance is associated with the particular call_id.

Third Party Call Control should not be used in conjunction with the PCOL feature.

Primary Rate Interface (PRI)

Primary Rate Interface (PRI) facilities may be used for either inbound or outbound adjunct-monitored calls.

An incoming call over a PRI facility provides the calling and called party information (CPN/BN/DNIS), which is passed on to the adjunct in the Call Offered to Domain Event Report and the Route capabilities as well as to events (for example, Alerting or Connected.)

An outgoing call over a PRI facility provides call feedback events from the network.

Switch-classified calls always use a call classifier on PRI facilities, whether the call is interworked or not. Although these facilities are expected to report call outcomes on the ‘‘D’’ channel, often interworking causes loss or delay of such reports. Progress messages reporting ‘‘busy,’’ SITs, ‘‘alert,’’ and ‘‘drop/disconnect’’ cause the corresponding event report to be sent to the adjunct. For switch-classified calls, the ‘‘connected’’ number is interpreted as ‘‘far end answer’’ and is reported to the adjunct as the Answered Event Report when received before the call classifier’s ‘‘answer’’ indication. When received after the call classifier has reported an outcome, it is not acted on. All outbound adjunct-monitored calls over PRI facilities do not generate the Trunk Seized Event Report. They may, however, generate the Alerting, Connected, Disconnect/Drop, and/or Call Ended Event Reports. If such a call goes ISDN end-to-end, other events are possible (for example, Alerting, Connected Event Reports). If such a call interworks, the PROgress message is mapped into a Cut-Through Event Report. In this case, only the Disconnect/Drop or Call Ended Event Reports may follow.

ASAI may provide UUI for inclusion in the Setup message for an outgoing ISDN-PRI call. This UUI can be provided in the ASAI Third Party Make Call, Auto Dial, or Route Select message. ASAI may receive UUI that was included in an incoming ISDN-PRI Setup message. This UUI will be included in any of the following messages, if they are sent for the call: Call Offered to Domain Event Report, Alerting Event Report, Route request.

ASAI may provide UUI for inclusion in an ISDN Disconnect message by including that UUI in the 3rd Party Drop message. ASAI may receive UUI that was included in an ISDN Disconnect message. The UUI would be included in a Drop event report, if one is sent to ASAI.

Priority Calling

Calls originated via the Third Party Auto Dial or the Third Party Make Call capability can be priority calls, if the adjunct specifies the *priority-call* option in the Third Party Auto Dial request. The user can also originate a priority call by going off-hook, dialing the feature access code for priority calling, and requesting a Third Party Auto Dial or a Third Party Make Call capability.

Privacy-Manual Exclusion

Activation of this feature affects all call control requests associated with other members of the bridge, TEG, or PCOL, if they try to originate on the bridged, TEG, or PCOL line appearance and affects the use of the analog principal's station if activated by a bridging user.

The exclusion feature can be activated associated with bridges, TEGs, or PCOLs. An exclusion button must be defined for the station that wishes to utilize this feature. Analog stations cannot utilize this feature since they do not have feature buttons. Activation of this feature when the station is a member of a bridge, TEG, or PCOL causes all other connected members of the group to transition to the bridged state. In addition, other members receive denial when they attempt to manually connect to the call.

Pressing the exclusion feature button toggles the feature from on to off as indicated by the green light associated with the button.

Ringback Queuing

Adjunct-routed or switch-classified calls are allowed to queue on busy trunks or stations.

When activated, the call back call reports events on the same call_id as the original call.

Send All Calls (SAC)

The adjunct can activate this feature by issuing a Request Feature capability. An adjunct can request a Third Party Make Call or Third Party Auto Dial capability for a station that has SAC activated.

A Third Party Auto Dial call to a station with the SAC feature active redirects to the covering station.

For incoming calls, the Alerting Event Report is sent only for multifunction sets receiving calls while having SAC activated. The Alerting Event Report is not generated for analog sets when the SAC feature is activated and the set is receiving a call.

Direct-agent calls existing in the agent queue when this feature is activated are not sent to coverage. Only the new direct-agent calls are affected by activation of this feature.

While SAC is active at a station, agent status value queries will show that an agent is busy.

SAC is ignored for priority direct-agent call and for switch-classified call destinations. If a SAC'ed call rings at a station with Domain Control, then the Alerting event includes a cause value indicating SAC as the reason the call redirected.

Service Observing

Domain-controlled stations may be service-observed and may also be observers. When a domain-controlled station is the observer, and is bridged onto a call for the purpose of service observing, the Connected Event Report is sent on domain-controlled associations for the observer's adjunct as well as to all other associations for that call.

Single-Digit Dialing and Mixed Station Numbering

A call initiated using the Third Party Auto Dial capability is permitted to use single digit dialing.

Station Message Detail Recording (SMDR)

Calls originated by the adjunct via the Third Party Auto Dial or Third Party Make Call capabilities are marked with the condition code ''B.'' Adjunct-originated calls include: calls originated by forcing the user off-hook after a Third Party Auto Dial or Third Party Make Call request; calls originated by the user going off-hook and then requesting Third Party Auto Dial or Third Party Make Call; and calls originated by the user going off-hook, dialing a few digits, and then requesting Third Party Auto Dial or Third Party Make Call.

Calls originated manually from a domain-controlled station are not marked with condition code ‘‘B.’’ Switch-classified calls are marked with condition code ‘‘B’’ and show either the ACD split or the agent’s extension (depending on how it has been administered) as the originator.

System Reports

The **list station** command indicates whether or not a given BRI link is ASAI.

Temporary Bridged Appearances

The operation of this feature has not changed with ASAI. No event is provided when a temporary bridged appearance is created at a multifunction set. If the user is connected to the call (becomes active on such an appearance), the Connected Event Report is provided. If a user goes on-hook after having been connected on such an appearance, a Disconnect/Drop Event Report with cause CS0/16 (normal clearing) is generated for the disconnected extension (bridged appearance).

If the call is dropped from the temporary bridged appearance by someone else, a Disconnect/Drop Event Report is also provided.

Temporary bridged appearances are not supported with analog sets. Analog sets get the Call Redirected Event Report when such an appearance would normally be created for a multifunction set.

The call state provided to queries about extensions with temporary bridged appearances is ‘‘bridged’’ if the extension is not active on the call or ‘‘connected’’ if the extension is active on the call.

The Third Party Selective Drop request is denied for a temporary bridged appearance that is not connected on the call.

Calls alerting at temporary bridged appearances may be redirected via Redirect Call. In this case the principal and temporary bridge will be dropped if redirection is successful.

Terminating Extension Group (TEG)

Members of a Terminating Extension Group (TEG) may be domain-controlled. A TEG behaves similarly to bridging for the purpose of ASAI event reporting. If controlled stations are members of a terminating group, an incoming call to the group causes an Alerting Event Report to be sent to all domain-control associations for members of the terminating group. On the domain-control association for the member of the group that answers the call, a Connected Event Report is returned to the answering member’s domain-control association(s) that contains the station that answered the call. All domain-control associations for the other group members (nonanswering members without TEG buttons) receive a Call Redirected Event Report. When a button TEG member goes on-hook but the TEG itself does not drop from the call, no event is sent but the state of that party changes from the connected state to the bridged state.

The Disconnect/Drop Event Report is not sent to each member's domain control associations until the entire TEG drops from the call (as opposed to an individual member going on-hook).

Members not connected to the call while the call is connected to another TEG member are in the bridged state. When the only connected member of the TEG transitions to the held state, the state for all members of the TEG changes to the held state even if they were previously in the bridged state. There is no event report sent over the domain-control associations for the bridged user(s) for this transition.

All members of the TEG may be individually domain-controlled. Each receives appropriate events as applicable to the controlled station. Call Control requests work normally if invoked over the station domain. However, Third Party Selective Hold, Third Party Merge, Third Party Reconnect, and Third Party Selective Drop are not permitted on parties in the bridged state and may also be more restrictive if the exclusion option is in effect from a station associated with the TEG.

Third Party Auto Dial or Third Party Make Call requests cannot specify the TEG group extension. TEGs can only receive calls, not originate them.

If a party_id query is requested while the TEG is alerting or on hold, one party member is reported for the group with the extension number specified as the TEG group extension (as the originator).

If a call query is requested on an extension while the TEG call is active, only one call appearance is associated with the particular call_id.

Timed Reminder

Third Party Auto Dial calls extended by an attendant and not answered redirect back to the attendant when the timed reminder interval expires. See ‘‘Attendant Call Waiting’’ earlier in this chapter for events returned to the adjunct.

Transfer

Manual transfer from a domain-controlled station is allowed subject to the feature's restrictions. The Hold Event Report is provided as a result of the first button push (or switch-hook flash for analog sets). The Call Transferred Event Report is provided as a result of the second button push (or on-hook for analog sets), and only if the transfer is successfully completed. The Transfer Event Report is sent to all active associations for the resultant call.

Trunk-to-Trunk Transfer

Existing rules for trunk-to-trunk transfer from a station user remain unchanged for adjunct-monitored calls. In such cases, transfers requested via Third Party Merge are negatively acknowledged (NAKed). When this feature is enabled, adjunct-monitored calls transferred from trunk to trunk are allowed, but there is no further notification (except for the Trunk Seized and Disconnect/Drop Event Reports sent to the adjunct).

Voice (Synthesized) Message Retrieval

A recording, ''Please call message center for more messages,'' is used for the case when the MWI has been activated by the adjunct through the Set Value capability.

CallVisor ASAI Over the DEFINITY LAN Gateway

13

Overview

A new transport option, CallVisor ASAI Over the DEFINITY LAN Gateway, is available in DEFINITY G3V4, and in DEFINITY G3V2/G3V3 with a field maintenance upgrade. This option incorporates a system assembly that uses a Multi-Function Board (MFB), a DEFINITY LAN Gateway circuit pack that supports an Ethernet controller and a software environment. The software environment, in turn, supports the DEFINITY LAN Gateway application that serves as an ISDN brouter of ASAI messages through a TCP ''tunnel'' via 10BaseT Ethernet.[1]

While CallVisor ASAI over the DEFINITY LAN Gateway supports the same Q.931 messages used for ASAI-BRI, it replaces the ISDN BRI transport layers below layer three with a simple TCP protocol described in the *DEFINITY Communications System CallVisor ASAI Protocol Reference*, 555-230-221.

1. In local area networking, a brouter is a device that combines the dynamic routing capability of an internetwork **router** with the ability of a **bridge** to interconnect dissimilar local area networks (LANs). It has the ability to route one or more protocols and bridge all other traffic. The DEFINITY LAN Gateway application links ISDN and TCP/IP at both a physical and addressing level.

Physical Connectivity

The system assembly that provides the DEFINITY LAN Gateway brouter application is a pair of circuit packs (a Multi-Function Board [TN2208] and an Ethernet Alarm Board [TN2170]), and a spacer, that provide a processor, hard disk, tape unit, Ethernet, and serial ports. See Figure 13-1.

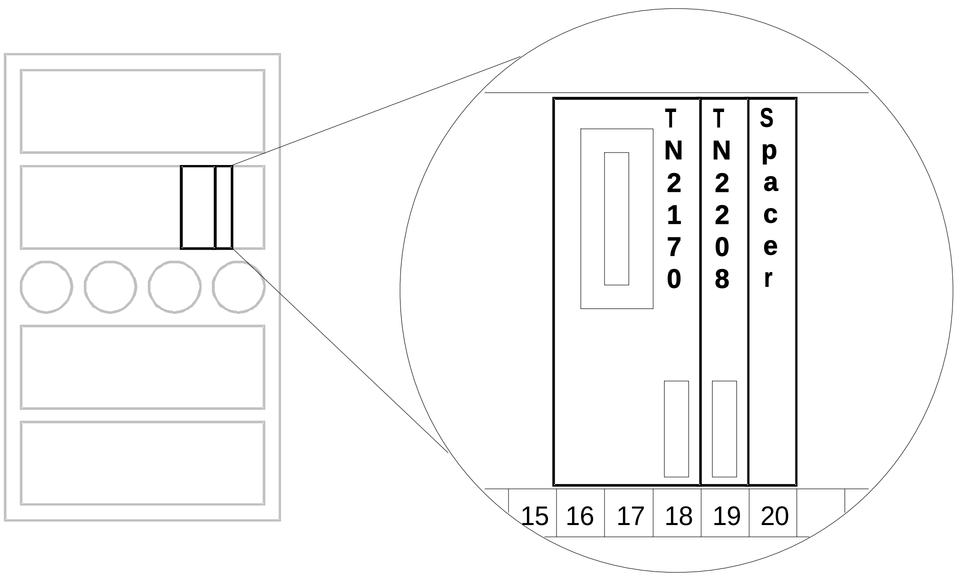


Figure 13-1. DEFINITY LAN Gateway System Assembly in a DEFINITY Carrier

The system assembly is inserted into a DEFINITY switch carrier using 5 contiguous slots. Once placed in the DEFINITY carrier, the system assembly is administered using the **change circuit-packs cabinet** command.

Carrier Connectivity

In DEFINITY G3V4 software releases (and in DEFINITY G3V2/V3 releases incorporating a field maintenance upgrade), the system assembly is recognized as the DEFINITY LAN Gateway application, and the **display circuit-packs cabinet** command shows the assembly occupying the selected slots.

For example, in Figure 13-1, the MFB occupies slots 16 through 20. Slot 19 should be administered as a DEFINITY LAN Gateway application on the MFB circuit pack (TN2208), and DEFINITY administration software will prevent slots 16, 17, 18, and 20 from being administered for other circuit packs.

However, if the system assembly is installed in a DEFINITY G3V3 or earlier, or in a DEFINITY G1, the circuit pack is recognized as a BRI circuit pack, and the **display circuit-packs cabinet** command shows only one of the physical slots as a TN556 (BRI) circuit pack. This is an unsupported switch configuration, and these switches should be upgraded to G3V4 or to a maintenance release of G3V3.

Cable Connectivity

The system assembly is provided with two special amphenol connectors, or cables, that are plugged into the back of the switch at the locations occupied by the TN2170 and TN2208 (see Figure 13-2). One cable has a female RJ45 receptacle that is used as the 10BaseT Ethernet connection. This cable is attached to the TN2170. The other cable consists of DB25 serial connectors and is attached to the TN2208. These cables provide Ethernet access and serial access to the DEFINITY LAN Gateway system assembly.

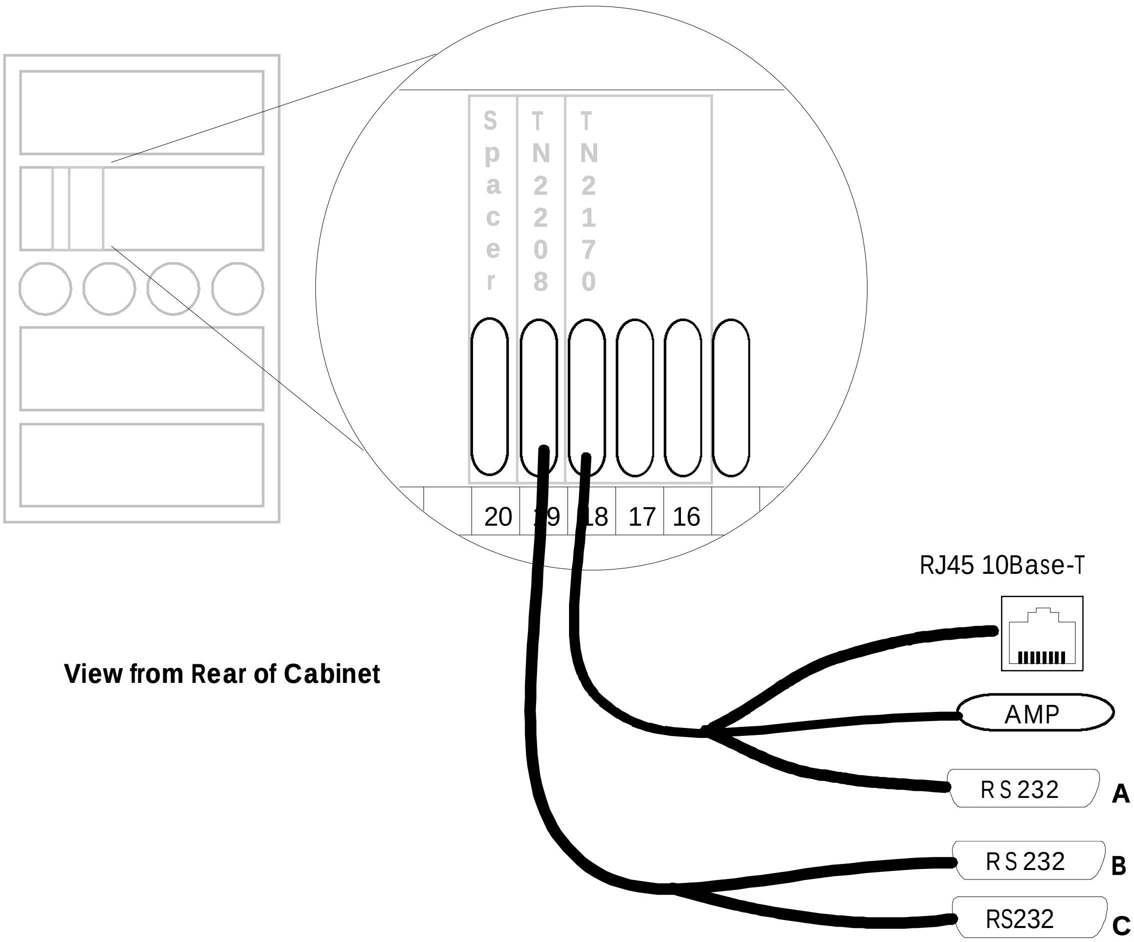


Figure 13-2. Cable Connectivity to the System Assembly

The DEFINITY LAN Gateway cable is attached to a ‘‘demarcation point’’ within 25 feet of the switch, consisting of a WE-104 terminal block with 2 RJ45 connectors punched-down back-to-back so as to provide a ‘‘straight-through’’ connection.

The terminal block, in turn, should be attached using suitable 10BaseT class 3 or better wiring to an Ethernet hub or an Ethernet hub adapter in the host to which physical connectivity is desired.

NOTE:
AT&T strongly recommends (both for security and performance reasons) that the Ethernet connectivity between the MFB and the set of hosts with which it will communicate be a separate LAN segment. Customers who do not follow this recommendation are subject to an unscrupulous person gaining access to the DEFINITY LAN Gateway application in order to commit toll fraud and/or tamper with the real-time aspects of CTI applications. (Toll fraud is the unauthorized use of your telecommunications system by an unauthorized third party. Under applicable law, the customer is responsible for paying for such unauthorized usage.)

The serial cable attached to the TN2170 (labeled A in Figure 13-2) should be connected to a 9600 baud (or better) modem that allows remote access by AT&T services. This is a similar arrangement to the INADS port provided on DEFINITY. To enhance customer security, the modem should be turned off except when access by AT&T is desired.

The serial cables attached to the TN2208 can be used for local administration, and **must** be used for initial configuration. These are RS232 ports that allow connectivity to a dumb terminal or to terminal emulation software on a PC. The amphenol connector on the cable attached to the TN2170 is not currently used.

Administrative Console Connectivity

The MFB is provided with two serial ports (labeled B and C in Figure 13-2) that provide access to the administrative logins. These ports may be connected to a dedicated terminal, or to a shared terminal through an ‘‘A-B box’’ arrangement. Additionally, once the LAN network administration is completed, the administrative application may also be accessed by the use of ‘‘telnet’’ from a remote system.

For customers who do not wish to have an additional, physical, permanent console, it is recommended that the MFB be initially administered using a temporary terminal, PC, or laptop. Once the LAN administration is completed, telnet may be used from any host attached to the isolated segment (such as the DEFINITY LAN Gateway client). This allows further (and ongoing) administrative access without the need for a permanent console terminal. The overall connectivity scheme is depicted in Figure 13-3 that follows.

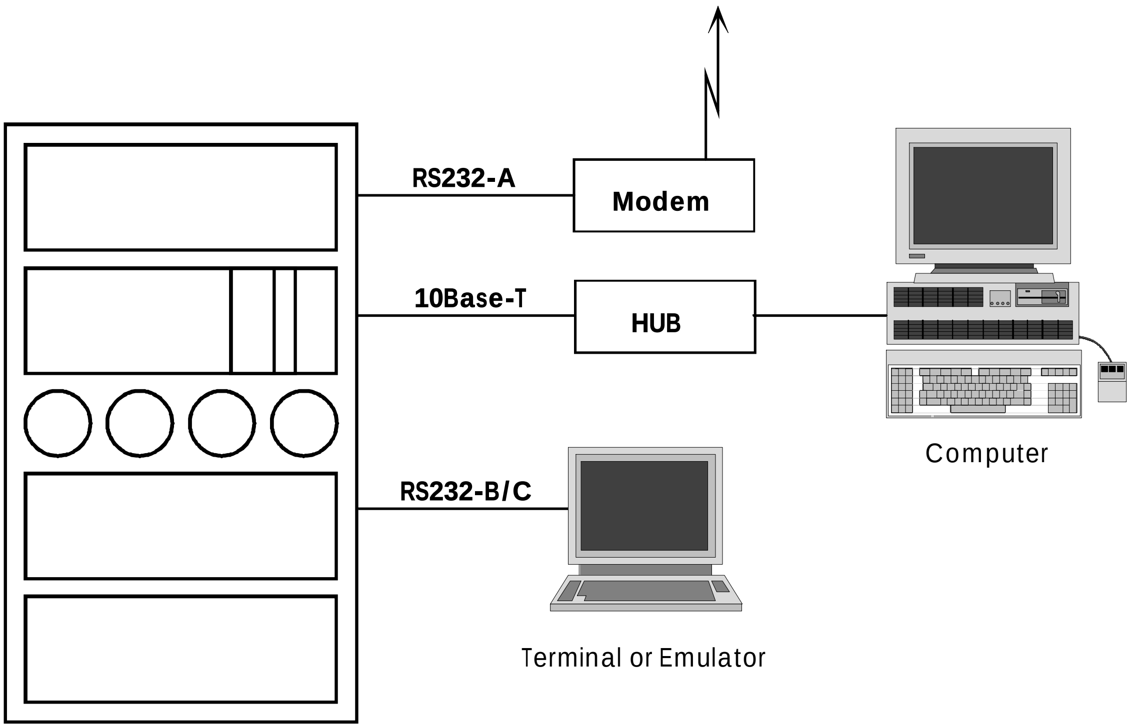


Figure 13-3. Overall System Connectivity

Administration

The DEFINITY LAN Gateway application may be administered using a terminal or terminal emulator with the RS232 console port, or using a terminal emulator that supports TCP connectivity using the LAN port. Initial administration, or administration to configure or diagnose LAN access on the MFB, should be performed using a serial port. Ongoing administration may be performed using either the serial or LAN port.

Administration is supported using a full-screen, menu-based application provided on the MFB and accessed via login and password. Administration screen categories are as follows:

- Login/Password Administration — Allows administrators to add and delete additional user logins. This screen also allows users to change their passwords.
- TCP/IP Administration — Allows administration of network parameters, including: network name and IP address of the DEFINITY LAN Gateway application, all locally known hosts, and all network routing information.

- Brouter Administration — Allows administration of the virtual-BRI-port-to-client-name/link table (see Chapter 7, ‘‘ASAI-Ethernet Protocol’’ in the *DEFINITY Communications System CallVisor ASAI Protocol Reference,* 555-230-221).
- Maintenance — Provides access to maintenance functions,
- Port Status/Control — Provides access to port status and control data. Allows administrators to view status information and terminate client connections.

These screens are located in *DEFINITY Communications System Generic 3 Installation, Administration, and Maintenance of CallVisor ASAI Over the DEFINITY LAN Gateway,* 555-230-223.

System Operation

ASAI is a point-to-point protocol. It does not include network addressing elements that indicate a particular client to which a message should be sent. Instead, a 1-to-1 correspondence between ASAI BRI ports and clients ensures that messages are sent to the proper destination. To support this, the DEFINITY LAN Gateway application creates a set of ‘‘virtual’’ BRI ports on the circuit pack. Thus, to support ASAI messaging on an Ethernet network it is necessary to ‘‘map’’ the virtual BRI ports to particular clients.

By performing this mapping, the system software performs a function similar to a LAN brouter. (Thus, the DEFINITY LAN Gateway software is referred to as a brouter.) It bridges ASAI messages from an ISDN/BRI synchronous point-to-point network to an Ethernet TCP/IP asynchronous network. The ASAI layer 3 messages remain the same; however, the system uses a TCP ‘‘tunnel’’ protocol for transport instead of ISDN layer 2.

Figure 13-4 illustrates the relationship among these elements.

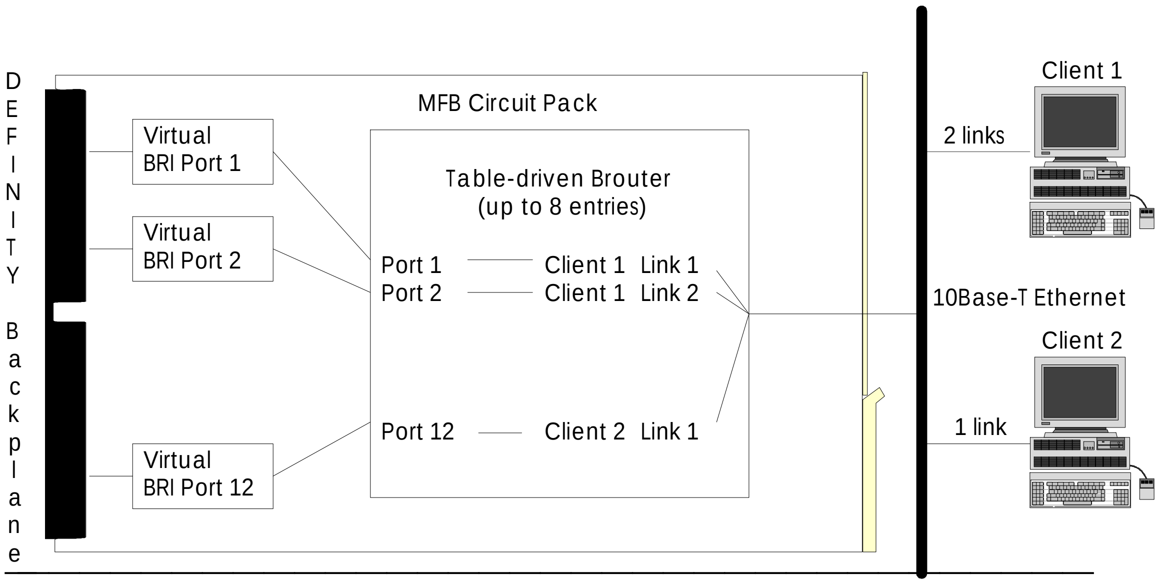


Figure 13-4. Relationship of Virtual BRI Ports, Brouter, and DEFINITY LAN Gateway Clients

Because of the point-to-point nature of ASAI, the brouter uses an administered table to determine the valid clients for CallVisor ASAI over the DEFINITY LAN Gateway. Each table entry provides a dedicated association between a client and a virtual BRI port. By using multiple table entries with different values for the client link, it is possible to provide a single client access to multiple virtual BRI ports. The DEFINITY LAN Gateway software represents itself to DEFINITY as a BRI circuit pack. This means that DEFINITY administration will continue to allow ‘‘ASAI’’ terminals to be assigned to BRI ports. The BRI ports may be actual, as they are with ASAI-BRI, or virtual, as they are using the DEFINITY LAN Gateway. If a BRI port is provided using the BRI circuit pack (TN556), then the BRI ports are actual. If a BRI port is provided using the DEFINITY LAN Gateway system assembly (TN2208 and TN2170), then the BRI ports are virtual and tunneled over Ethernet TCP/IP.

Once a virtual BRI port is assigned in the brouter application, all its ASAI messages are sent to the administered client, if it is connected. The brouter also passes ASAI messages from DEFINITY LAN Gateway clients to the virtual BRI port and then to the DEFINITY switch.

To pass messages, the brouter references the table of virtual BRI ports, client host names, and link numbers. A maximum of 8 such connections or mappings can exist simultaneously (only 4 are useful in a DEFINITY G3s). Each entry in the table has the following form:

Table 13-1. Brouter Table Format

Client Name or IP Address	Client Link Number	DEFINITY BRI Port Number

These entries are explained as follows:

- Client Name or IP Address — The host name or IP address of the client authorized to use the specified DEFINITY BRI port. If a client host name is provided, then a user must ensure that the host table can resolve the host name.
- Client Link Number — The link number the client will use when attempting to connect to the brouter. The valid range is 1 to 8. This parameter is used to distinguish between multiple links assigned to a single client.
- DEFINITY BRI Port Number — The number of the DEFINITY virtual BRI port used for this client's requested link. The valid range is 1 to 12.

The DEFINITY LAN Gateway software is shipped from the factory with a default IP address of 192.168.25.10 and a default host name ‘‘definity.’’ It is also shipped with a default **client** IP address of 192.168.25.20 and hostname ‘‘client.’’ The brouter listens for connections from clients on TCP port number 5678. The client must establish a TCP connection to the brouter at this port and IP address. The customer may change the IP address and/or hostname, but the TCP port is fixed.

For more information on CallVisor ASAI Over the DEFINITY LAN Gateway, see the *DEFINITY Communications System Generic 3 Installation, Administration, and Maintenance of CallVisor ASAI Over the DEFINITY LAN Gateway,* 555-230-223.

Installation and Test for CallVisor ASAI

14

This is also chapter 2 in the *AT&T DEFINITY Communications System Generic 3 CallVisor ASAI Planning Guide,* 555-230-222.

Hardware Installation

The CallVisor ASAI link requires the following hardware components:

1. ISDN-BRI Circuit Pack (for CallVisor ASAI-BRI link) or ASAI-Ethernet Multi-Function Board (for CallVisor ASAI-Ethernet link). (Multiple links may use both.)
2. Packet Controller Circuit Card (for internal communications)
3. Packet Maintenance Circuit Pack

Check that your DEFINITY Communications System has the above components. If installation of any of the above components is required, refer to *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test*, 555-230-104.

The following hardware may be needed for your specific CallVisor ASAI application. See *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test*, 555-230-104, for complete information on how to install and test the hardware.

- Announcement Circuit Pack — For integrated announcements
- Call Classifier Circuit Pack — For predictive dialing call classification or for call prompting applications
- DS1 Interface Circuit Pack (ISDN) — For CPN/BN delivery

- Facility Test Circuit Pack — Required for duplicated systems only; optional for unduplicated
- Expansion Interface Circuit Pack — For new multiple-port network systems or upgrades. (If you have the older TN776 circuit pack, it must be replaced with a TN570 circuit pack or later, which permits CallVisor ASAI to terminate on an Expansion Port Network [EPN]).
- MultiQuest trunk for Flexible Billing

Software Installation

In addition to the CallVisor ASAI and the DEFINITY Communications System basic software, the following software might be needed for your specific CallVisor ASAI installation. Refer to the *DEFINITY Communications System Generic 1 and Generic 3 Installation and Test*, 555-230-104, for more information.

- Automatic Call Distribution (ACD) Software — For ACD configurations
- Automatic Route Selection (ARS) Software — For ISDN-PRI call routing
- ISDN-PRI Software — For ISDN-PRI trunks
- Vectoring Basic Software — For call vectoring applications and ASAI adjunct routing
- Vectoring Prompting Software — For integrated call prompting applications
- Vectoring (ANI II Digits Routing)
- Vectoring (G3V4 enhanced)
- Vectoring (G3V4 advanced routing)
- EAS Agent Skills
- Answering Machine Detection for Predictive Outbound Call Management (OCM)
- AT&T Adjunct Links
- VDN Return Destination
- Optionable Switch-Classified Calling
- AT&T MultiQuest Flexible Billing
- ASAI-Accessed Internally Measured Data
- VuStats

CallVisor ASAI Link Administration

The CallVisor ASAI software must be installed in your system before an ASAI link can be administered and activated. Up to eight ASAI links can be administered at the same time to the same or different adjunct processors, and using either BRI or Ethernet transport, or both.

To administer an ASAI link, use the **add station** command and set the link type to `ASAI`. Fill out the remainder of the form as appropriate for your ASAI host. CallVisor ASAI links do not support Management Information Messages (MIMs). Screen 2-1 shows a sample completed Add Station form.

```
 add station                                                       Page 1 of 1
-----------------------------------------------------------------------------
                               STATION

Extension: 5000       BCC: 0
     Type: ASAI                                         COR: 1
     Port: 1A0102                                       COS: 1
     Name: asai link

EVENT MINIMIZATION? n


                      XID? y                 Fixed TEI? y   TEI: 3
              MIM Support? n
                                            CRV Length: 2
```

Screen 14-1. Add Station

For more information on CallVisor ASAI link administration, see the *DEFINITY Communications System Generic 3 CallVisor ASAI Protocol Reference,* 555-230-221, and the *DEFINITY Communications System Generic 3 Feature Description*, 555-230-204.

Table 14-1 shows how feature options are specified for different ASAI vendor partners' equipment. `MIM Support?` must always be set to no.

Table 14-1. ASAI Feature Options Administration for AT&T Vendor Partners (ISDN Links Only

Vendor Partner	XID	MIM Support	Fixed TEI	CRV Length	Event Min.
AT&T CONVERSANT® VIS 4000	Yes	No	Yes,TEI=3	1 byte	n
AT&T CONVERSANT VIS 5000	Yes	No	Yes,TEI=3	2 bytes	n
Dialogic CT-Connect™	No	No	Yes, TEI=55	2 bytes	n
International Business Machines (IBM®)	No	No	No	2 bytes	y
Stratus Computers	No	No	Yes,TEI=1	2 bytes	n
Aristacom	No	No	No	2 bytes	n
Hewlett-Packard	Yes	No	No	2 bytes	n
CallVisor PC (including EIS)	Yes	No	Yes,TEI=3	2 bytes	n
CallVisor 3000	No	No	No	2 bytes	n
Novell	Yes	No	Yes,TEI=3	2 bytes	n
IBM ASAI-Ethernet clients	No	No	Yes,TEI=3	2 bytes	y
All other ASAI-Ethernet clients	No	No	Yes,TEI=3	2 bytes	n

See ''AT&T Vendor Partners'' that follows in this chapter for more information.

In addition to the CallVisor ASAI link administration, the DEFINITY Communications System features that follow may need to be administered for your specific ASAI application. The required forms are listed as well.

- Adjunct-Controlled Splits — Add/Change Hunt Group Form (For the majority of applications, this field does not apply and should be left at the default value of NONE.)
- ASAI Adjunct Alarm Administration — The system default for CallVisor ASAI alarms is to provide warnings when an ASAI link fails or the adjunct does not respond to switch messages. The customer can tune the level of on-board and off-board alarms to warnings, or minor or major alarms. Alarm severities are set by means of the **set options** maintenance command, explained in *DEFINITY Communications System Generic 3r Maintenance*, 555-230-105. Once the ASAI adjunct alarms are

administered to an alarm level, the switch alarms all CallVisor ASAI links at that level.

- Automatic Call Distribution (ACD) — Add/Change Hunt Group, Trunk Group, Station, Feature-Related System Parameters, and Attendant Console Forms
- EAS Skills and Logical Agents — Agent Login ID form, Add/Change Hunt Group form, and Add/Change VDN form
- Automatic Route Selection (ARS) — Change ARS Digit Analysis Form
- Call Vectoring — Add/Change VDN and Call Vector Forms
- ISDN-PRI Trunks and Options — Add/Change Trunk Group Form
- Outbound Call Management (OCM) Special Information Tones (SITs) — Remote Access Form
- Answering Machine Detection — System-Parameters Customer-Options Form
- Option for Switch-Classified Calls — System-Parameters Customer-Options Form
- Flexible Billing — System-Parameters Customer-Options Form. In addition, the following optional system features must be activated to support the MultiQuest Flexible Billing feature:
  - ASAI
  - ISDN — Primary Rate Interface (ISDN-PRI)
  - MultiQuest Flexible Billing

  Finally, the following optional system features will probably be used when Flexible Billing is active:
  - Automatic Call Distribution (ACD) — Required if call routing uses ACD
  - Call Vectoring — Required if inbound calls use Call Vectoring for routing
- ASAI-Accessed Internally Measured Data — System-Parameters Customer Options Form. In addition, the following system features must be activated to support the ASAI-Accessed Internally Measured Data feature:
  - Internally Measured Data option
  - VuStats

For detailed instructions regarding the administration of the above features, see the appropriate (G3i or G3r) *DEFINITY Communications System Generic 3 Implementation* document, 555-230-650 or 555-230-651, respectively.

CallVisor ASAI Link Testing

After the CallVisor ASAI link is administered and installed, perform the following tests as described in the *DEFINITY Communications System Generic 3 Installation and Test*, 555-230-104.

- Status Station
- Status bri-port
- Test port

In addition, check that the switch does not report any alarms or errors associated with the CallVisor ASAI port.

NOTE:
AT&T tests that the CallVisor ASAI link is functional as part of the basic installation procedures. Coordination of additional testing can be done by AT&T if the Single Point of Contact Enhanced Installation Offering is purchased by the customer. Actual testing of other vendors' equipment and applications is performed by the customer or appropriate representative.

Additional testing of the adjunct application must be performed by the customer to ensure that the application performs as desired.

AT&T Vendor Partners

The following AT&T vendor partners have signed development agreements with AT&T to share information about each other's products and to establish working relationships among companies.

NOTE:
This list of vendor partners is up-to-date at the time of this document's publication. However, the number of vendor partners may expand in the future. For a current list and more detailed information, contact your AT&T representative.

- Dialogic CT-Connect
- International Business Corp. (IBM)
- Hewlett-Packard (HP)
- Aristacom
- Stratus

- Tandem
- Novell (PassageWay™ Telephony Services)

> **NOTE:**
> This product is known as NetWare® Telephony Services if sold by Novell instead of AT&T.

- Electronic Information Systems (EIS)

See Appendix E in the *AT&T DEFINITY Communications System Generic 3 CallVisor ASAI Planning Guide*, 555-230-222, for CallVisor ASAI functionality supported by AT&T vendor partners.

Call Scenarios and Applications

A

This Appendix has the following sections:

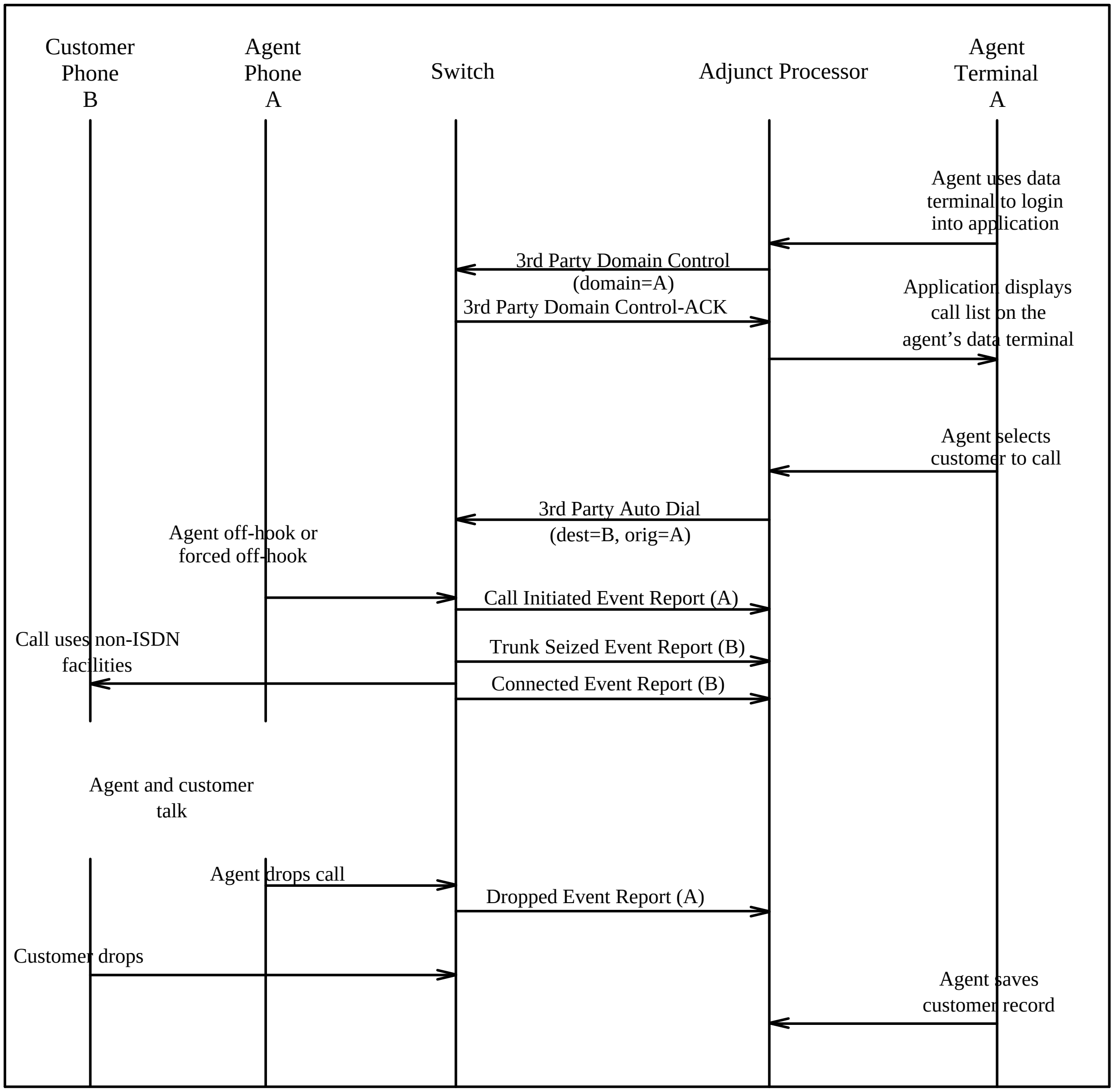
1. Basic Application Call Scenarios
2. Calls Across Multiple Switches
3. Expert Agent Selection Interactions
4. Converse Vector Command Interactions
5. Redirection on No Answer (RONA) Interactions
6. VDN in Coverage Path Interactions
7. User Scenarios — User to User (UUI) Information
8. User Scenarios — Connected IE for Non-ISDN Trunks
9. User Scenarios — ASAI-Provided Dial-Ahead Digits
10. User Scenarios — ASAI-Requested Digit Collection
11. User Scenarios — VDN Return Destination
12. ASAI Messaging Scenarios — VDN Return Destination
13. User Scenarios — Flexible Billing

1. Basic Application Call Scenarios

The following call scenarios show how ASAI capabilities can be used to implement several ASAI applications. Only sample parameters are given with the ASAI messages; not all parameters are provided. For example, the messages do not show call identifiers or party identifiers for each message. Duplicate event reports and the associations that correspond to each message are not shown either.

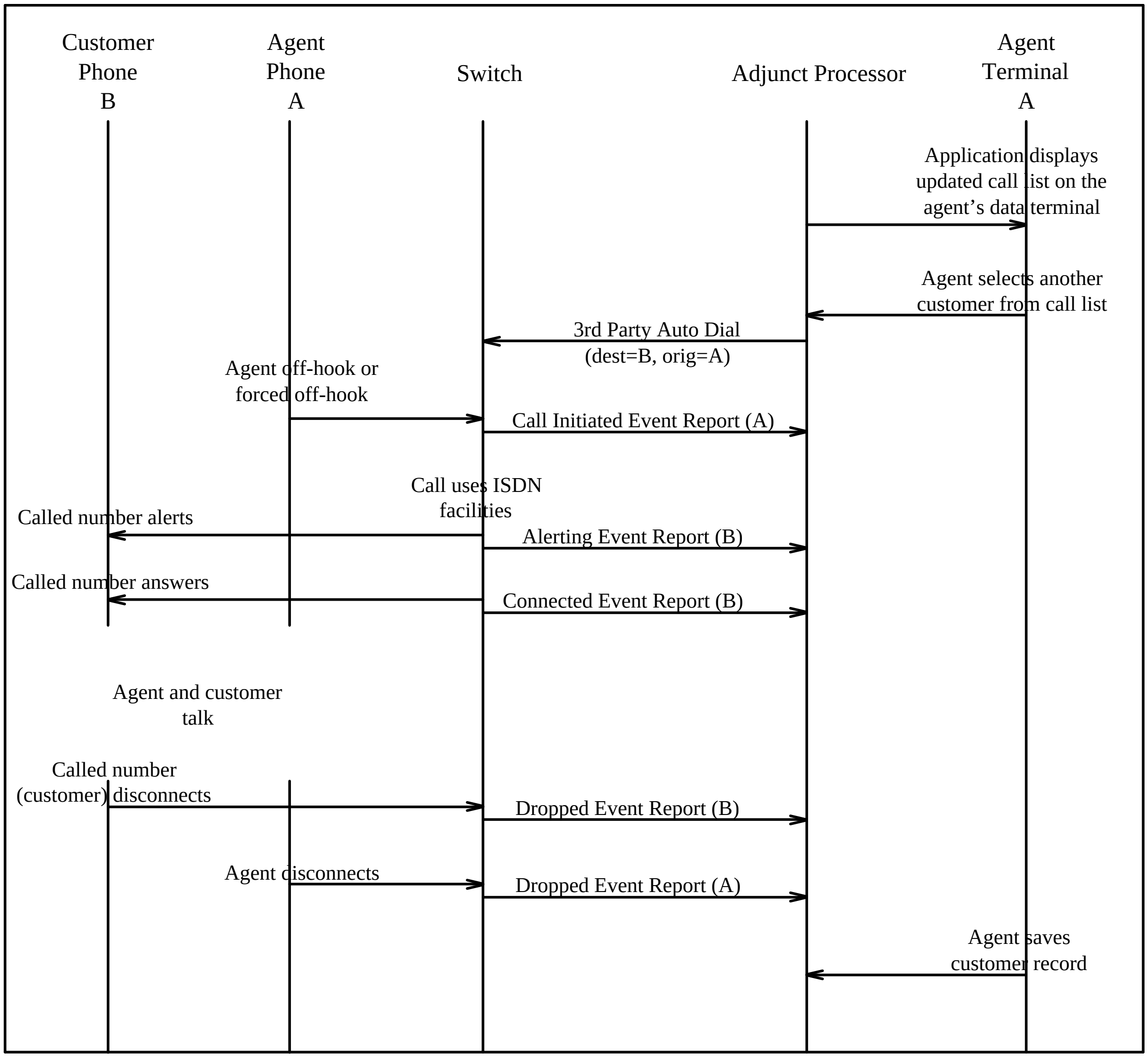
The application is assumed to know the telephone or extension number associated with each VDN, ACD split, and agent or user shown. Additional assumptions are provided with each sample scenario.

Outbound Call Management — Preview Dialing (Non-ISDN Facilities)



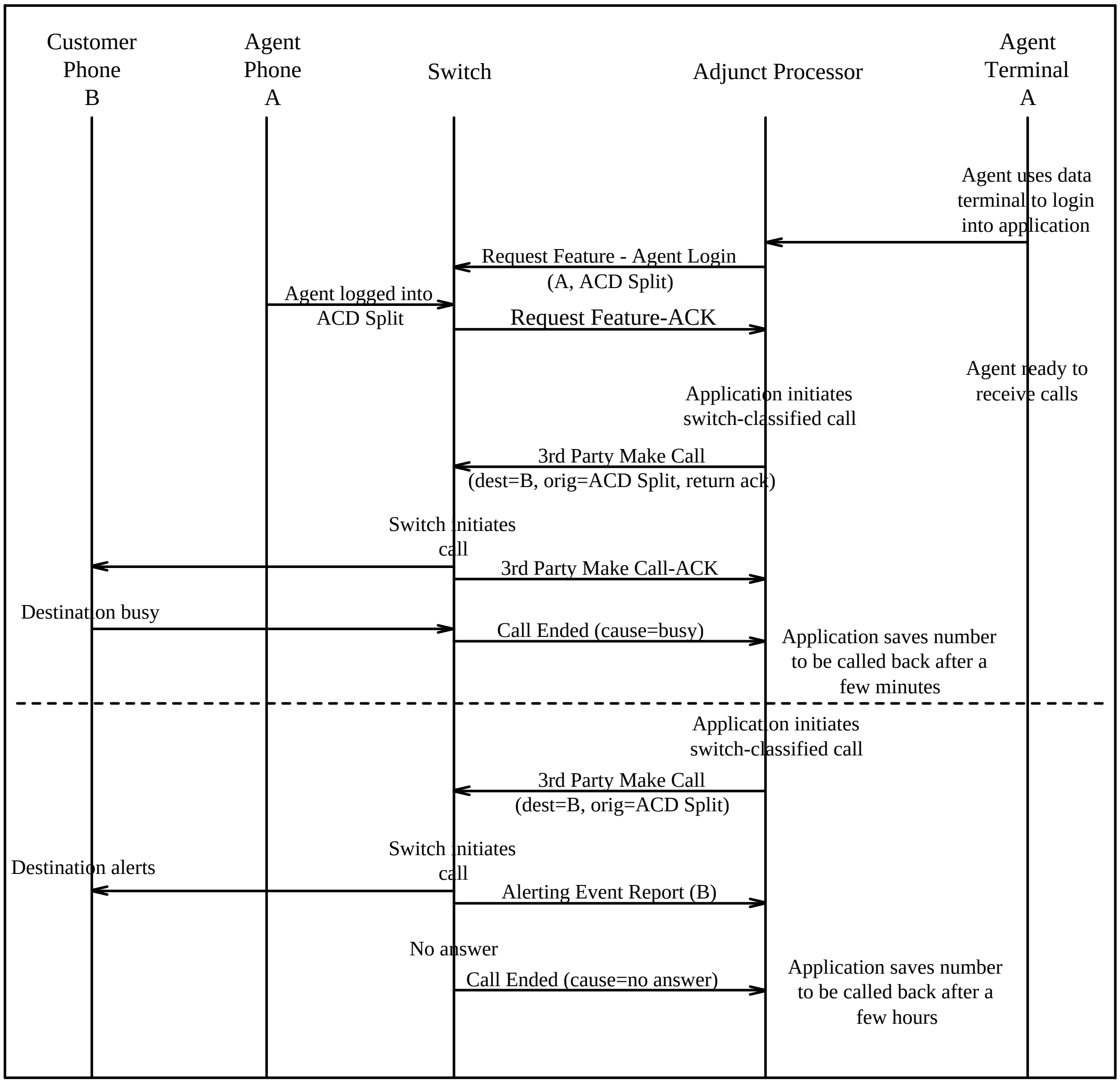
The application in the adjunct processor provides the agent and/or user with a call list. The agent uses the data terminal to select the destination to call. The application then requests a call on behalf of the agent. The call uses non-ISDN facilities.

Outbound Call Management — Preview Dialing (ISDN Facilities or Local Extensions)



The application in the adjunct processor provides the agent and/or user with a call list. The agent uses the data terminal to select the destination to call. The call uses ISDN facilities. Local destinations (that is, switch extensions) provide the same event reports as the one presented for ISDN facilities.

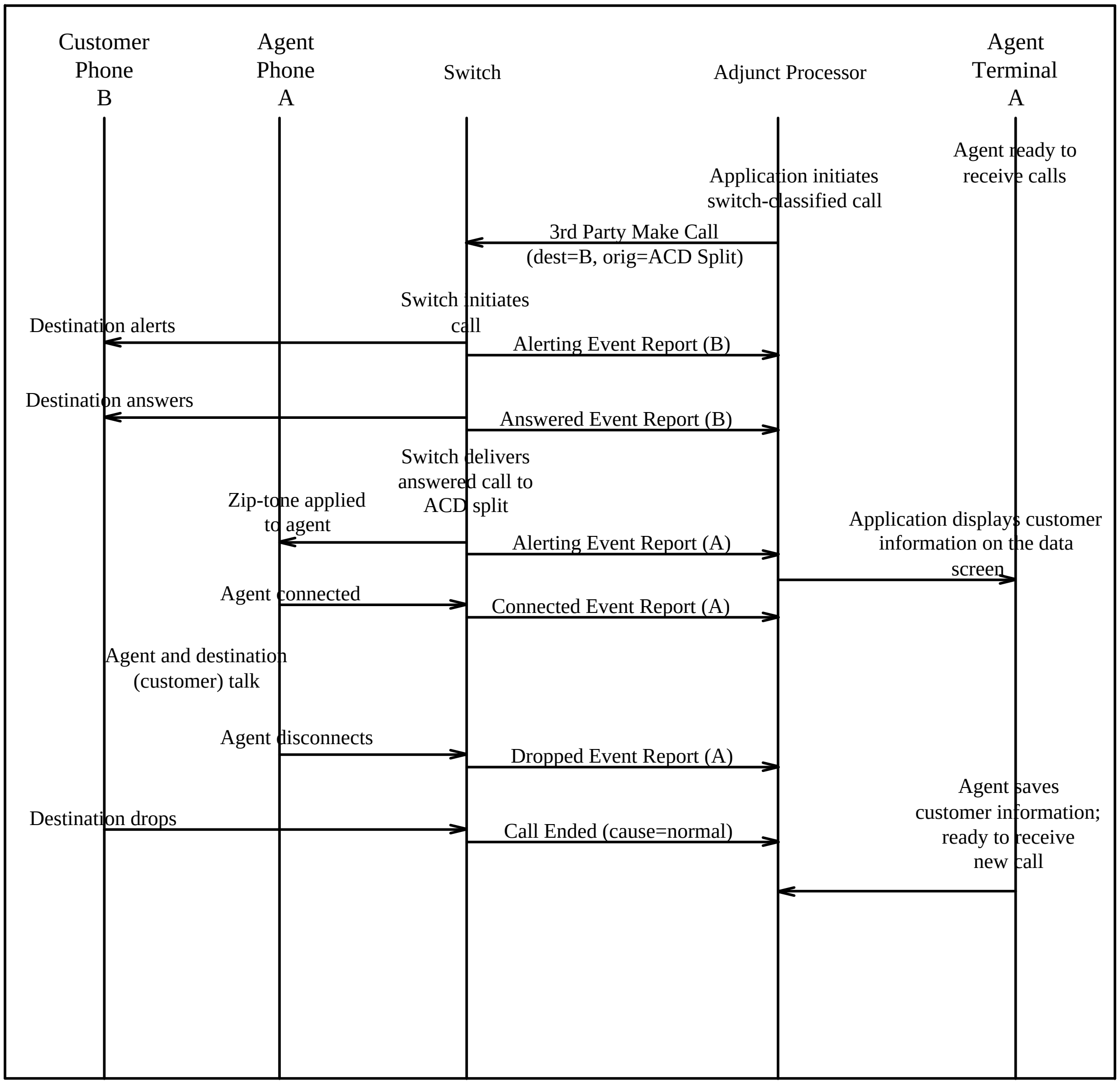
Outbound Call Management — Predictive Dialing (Destination Busy and No Answer)



The application in the adjunct processor requests **Third Party Make Calls** with the **alert_dest_first**, **service circuit = switch-classified**, and **max_ring_cycles** options set (that is, switch-classified calls). The first call shown receives busy tone. The second call is not answered within the specified **max_ring_cycles** time.

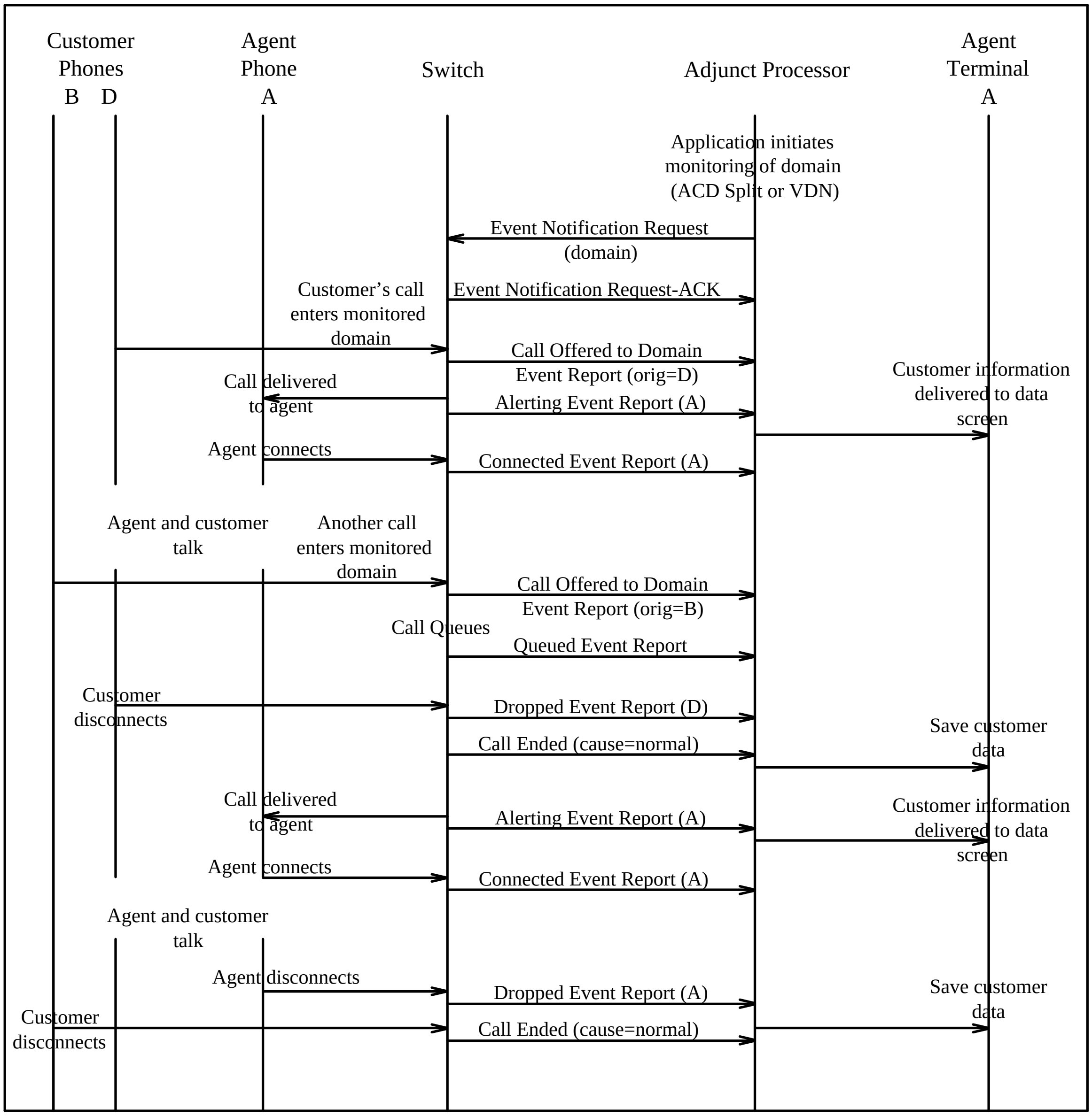
Note that the agent is not notified of calls that receive busy or are not answered. The application stores the call outcomes (for example, busy, no answer, SIT tone) for later call processing.

Outbound Call Management — Predictive Dialing (Success)



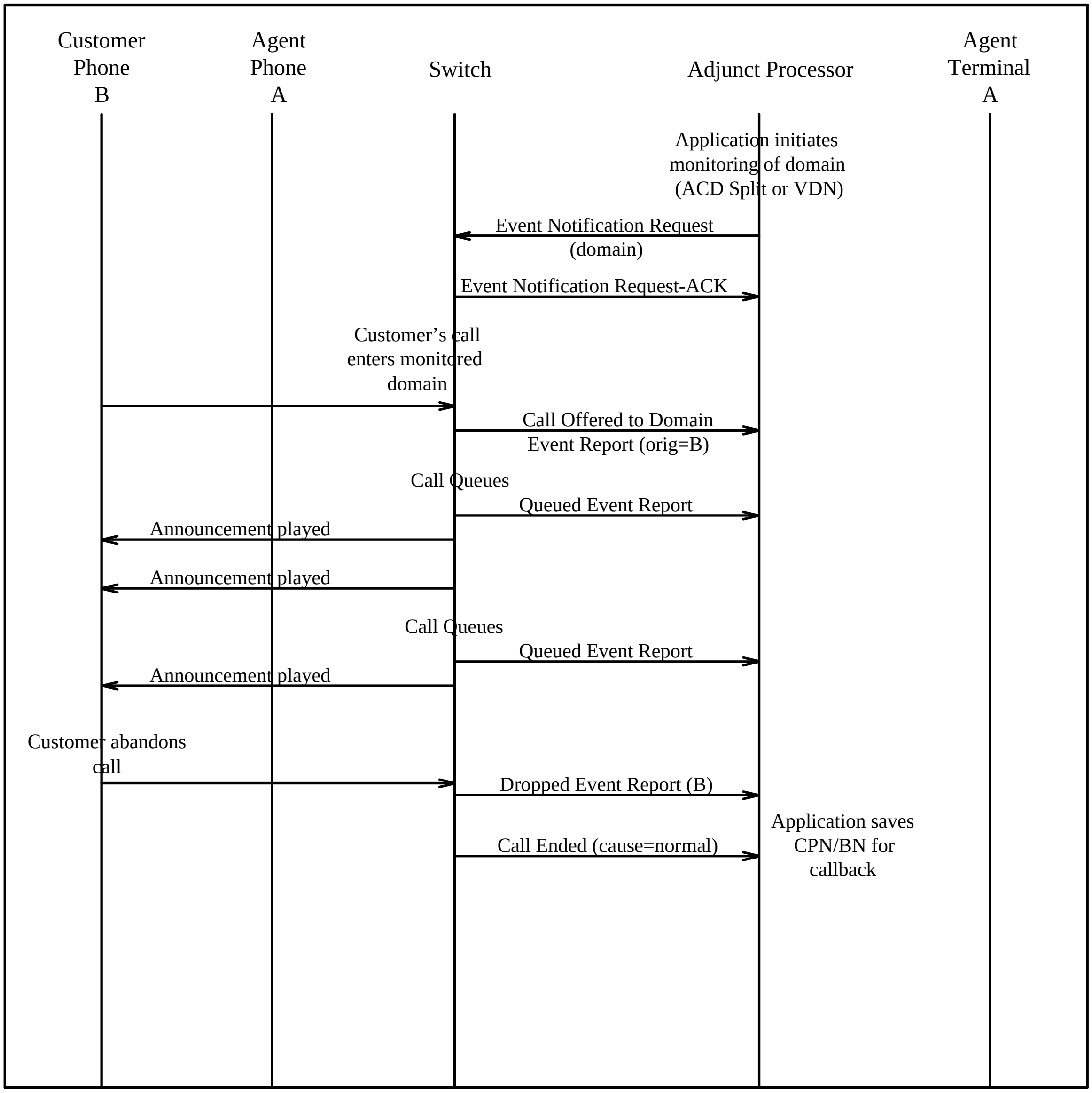
The application in the adjunct processor requests **Third Party Make Calls** with the **alert_dest_first**, **service circuit = switch-classified**, and **max_ring_cycles** options set (that is, switch-classified calls). The call is answered and delivered to the originating ACD split. The ACD split immediately delivers the call to an available agent. If no agent is available, the call can be diverted to an announcement.

Call Monitoring — VDNs and ACD Splits



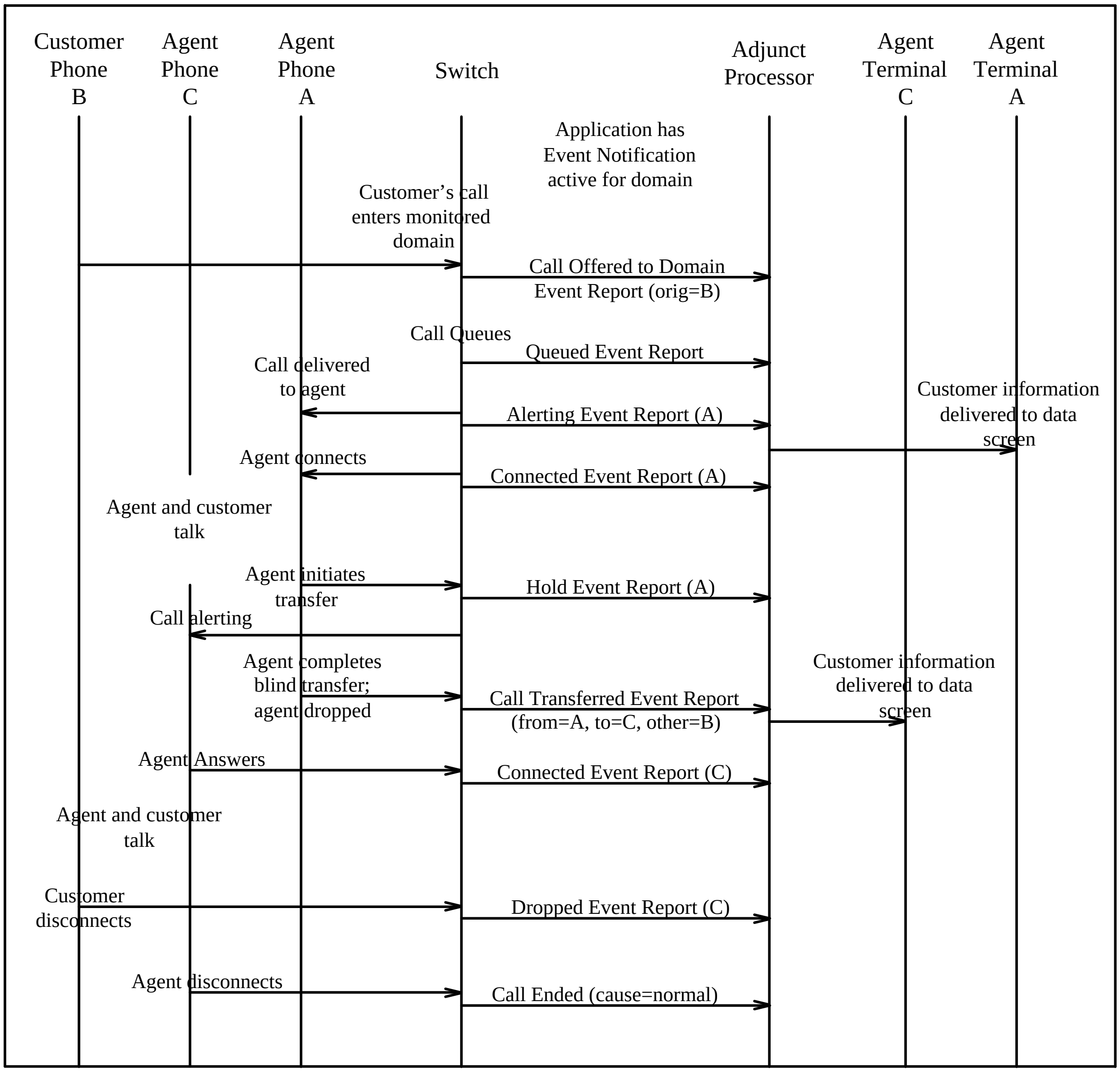
The application in the adjunct processor monitors all calls entering a VDN or ACD split. All calls have a unique call identifier (call_id) that the adjunct/application uses to track calls (call identifiers are not shown). The first call that enters the monitored domain does not queue, since there is an available agent to answer the call. The second call that enters the monitored domain waits in queue until agent A transfers the first call (destination not relevant to this example). Note that despite the first call being transferred to another destination, the application continues to receive event reports for the call.

Multiple Queuing and Call Abandon— ACD Split or VDN Monitoring



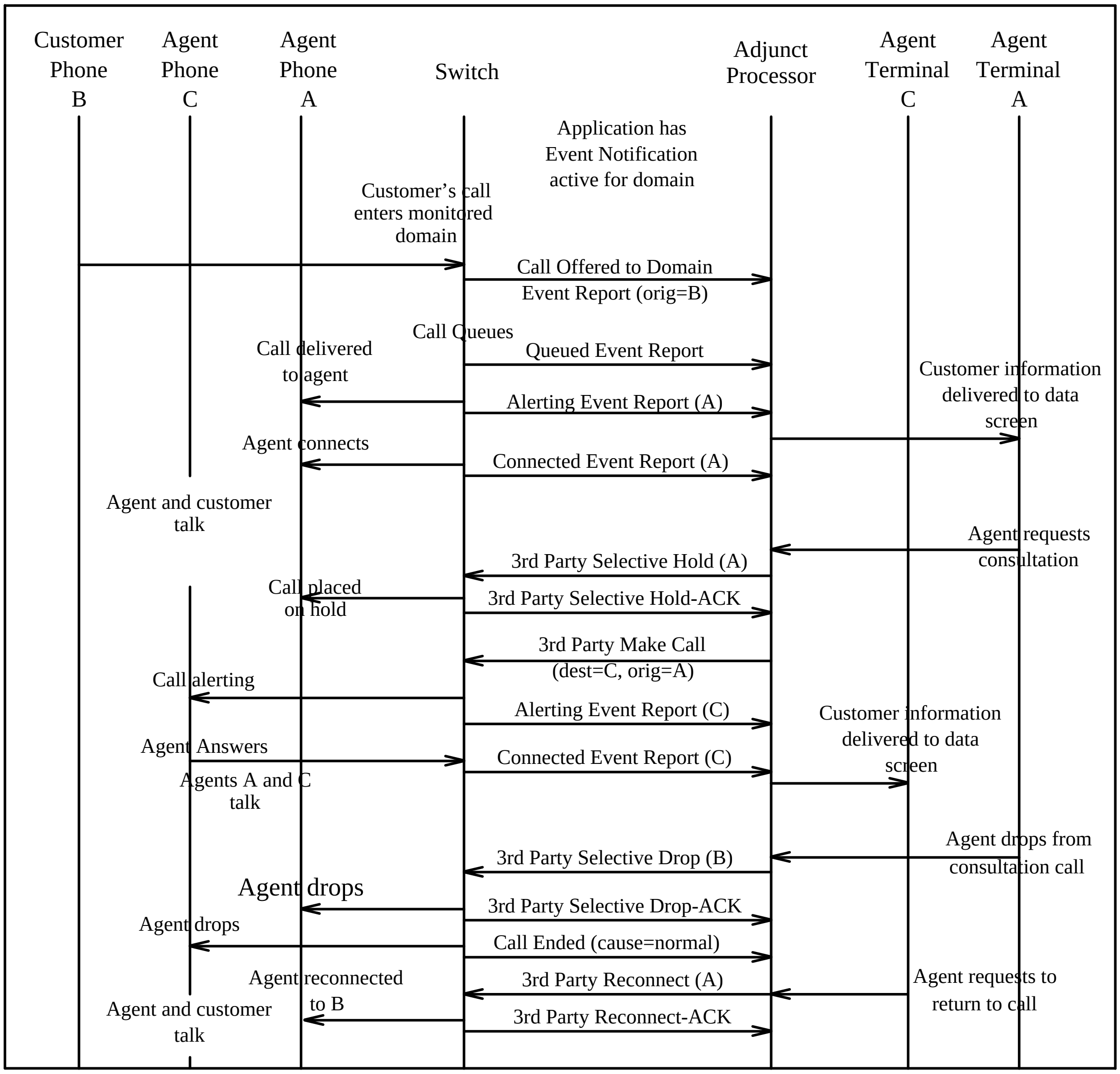
The application at the adjunct processor monitors all calls entering a VDN or ACD split. This scenario shows a call receiving multiple announcements and queuing into different splits. The customer disconnects before he or she is connected to an agent.

Blind Transfer — ACD Split or VDN Monitoring



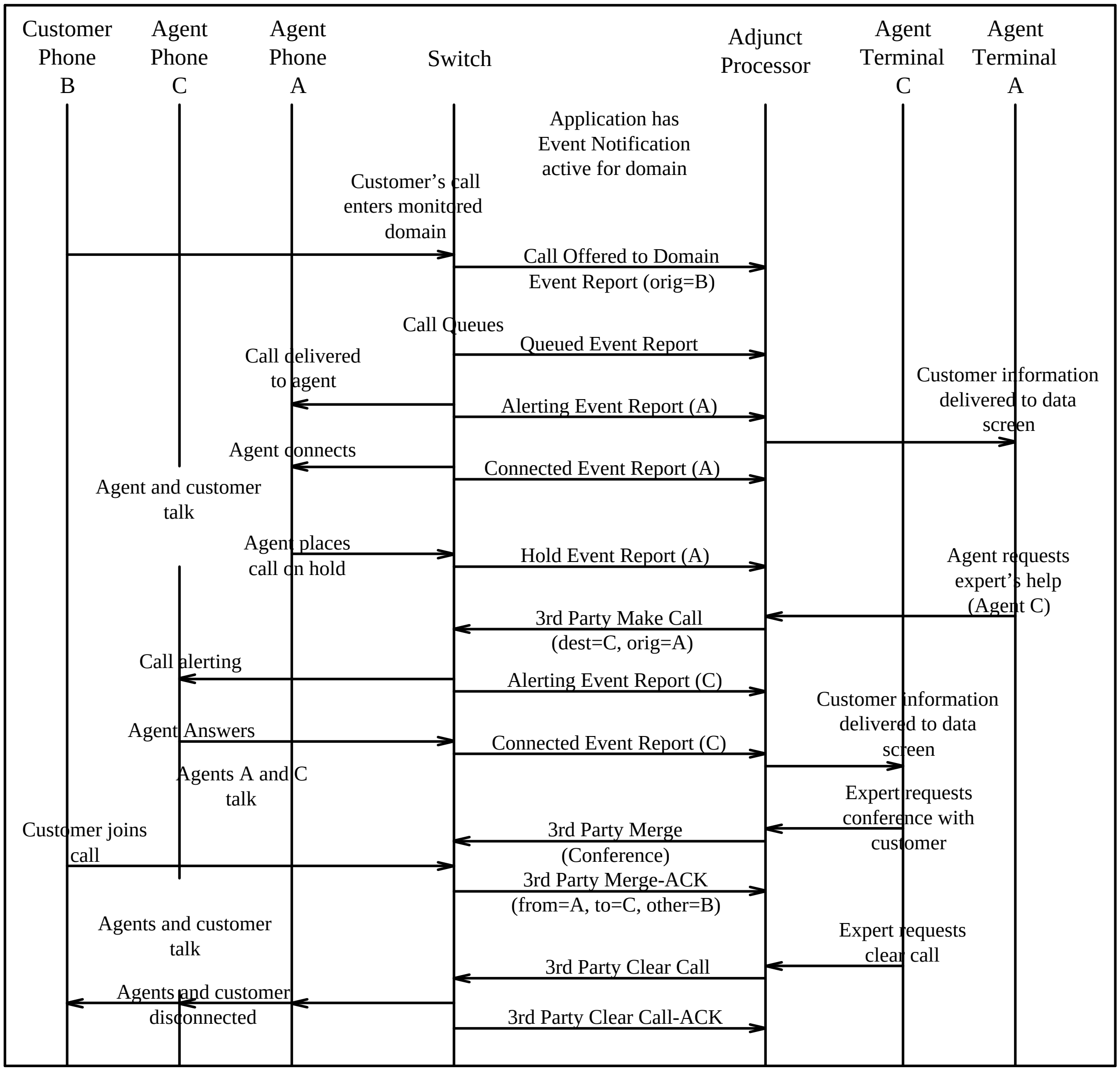
The application at the adjunct processor monitors all calls entering a VDN or ACD split. This scenario shows agent A performing a blind transfer to agent C. The transfer operation is done at the agent's station by pressing the transfer button, dialing the second call, and pressing the transfer button a second time. The adjunct processor does not control the transfer operation.

Consultation — ACD Split or VDN Monitoring



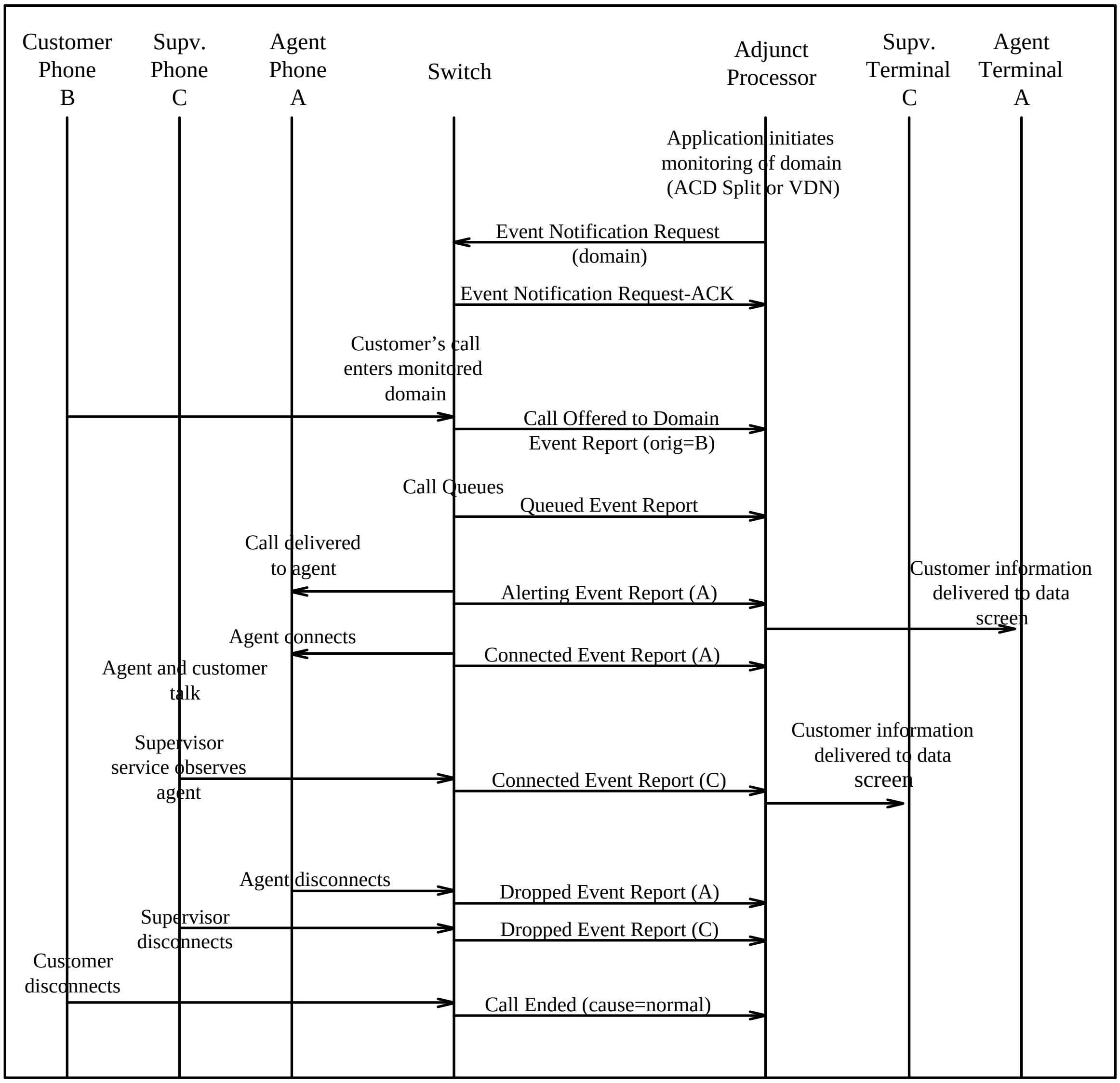
The application at the adjunct processor monitors all calls entering a VDN or ACD split. This scenario shows an ACD agent placing a customer's call on hold and consulting with another agent. The agent initiates the consultation call and the reconnection to the customer via the data terminal. The **Third Party Selective Hold** and **Third Party Reconnect** operations assume that the application has taken control (**Third Party Take Control** operation not shown) of the call. The Hold and Reconnected Event Reports for the Event Notification association are not shown either.

Agent Conference



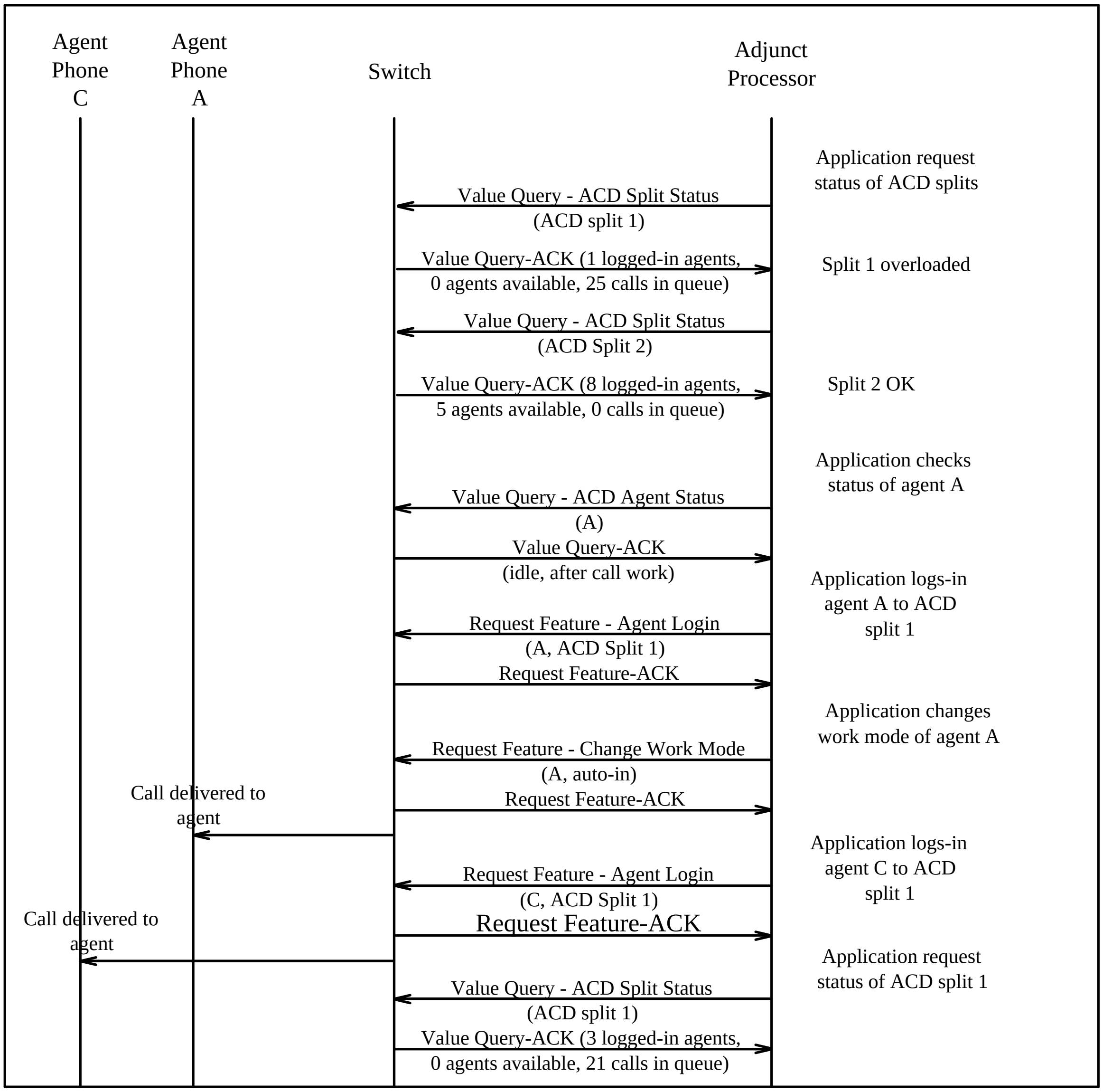
Agent A performs a conference operation from the data terminal by using the **Third Party Make Call** and **Third Party Merge** capabilities. In addition, the agent uses **Third Party Clear Call** to terminate the call.

Service Observing — ACD Split or VDN Monitoring



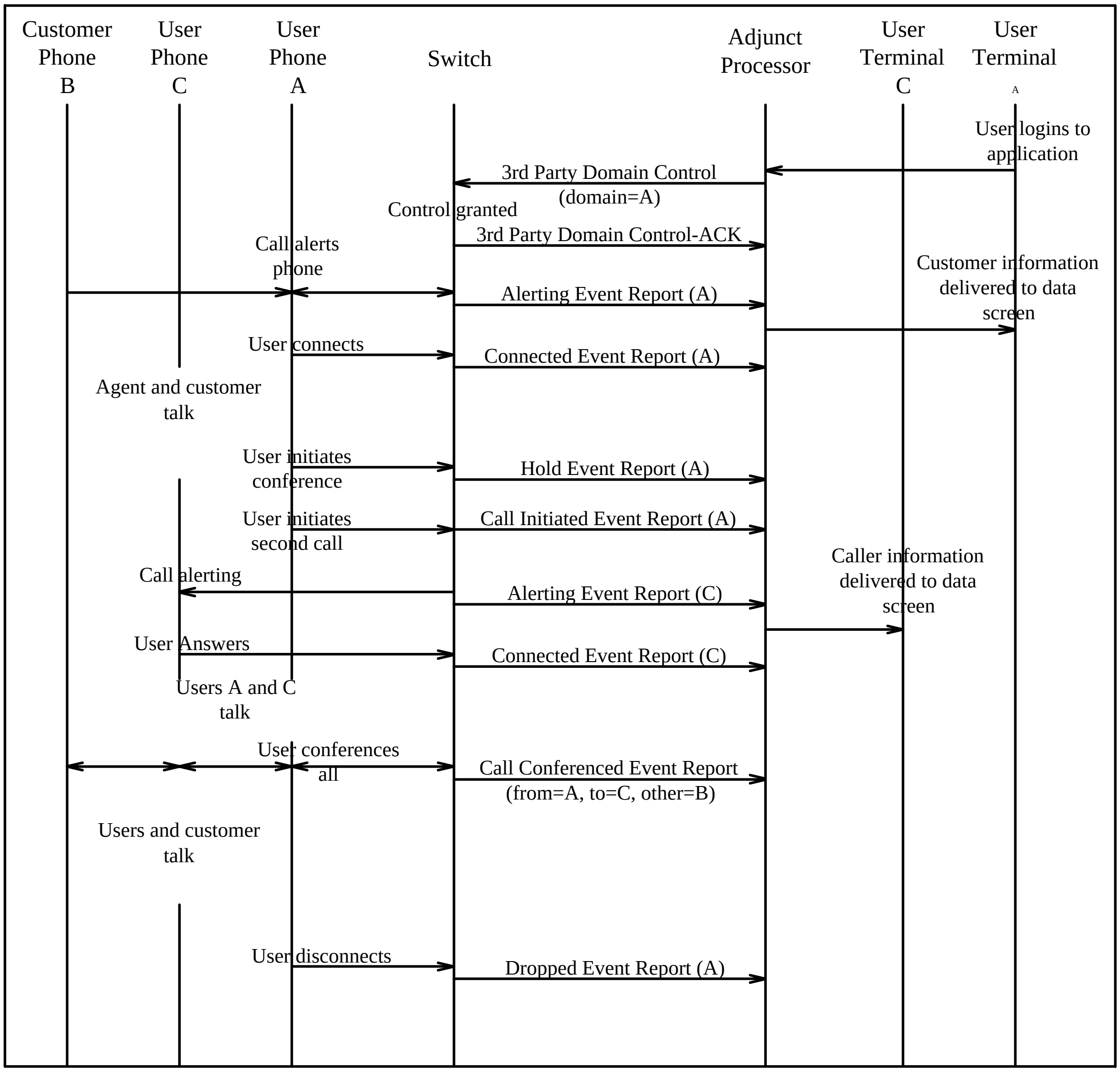
A supervisor service-observes an agent that receives calls monitored by the application in the adjunct processor. The service observing operation is requested manually by the supervisor at his or her station.

Agent Reconfiguration



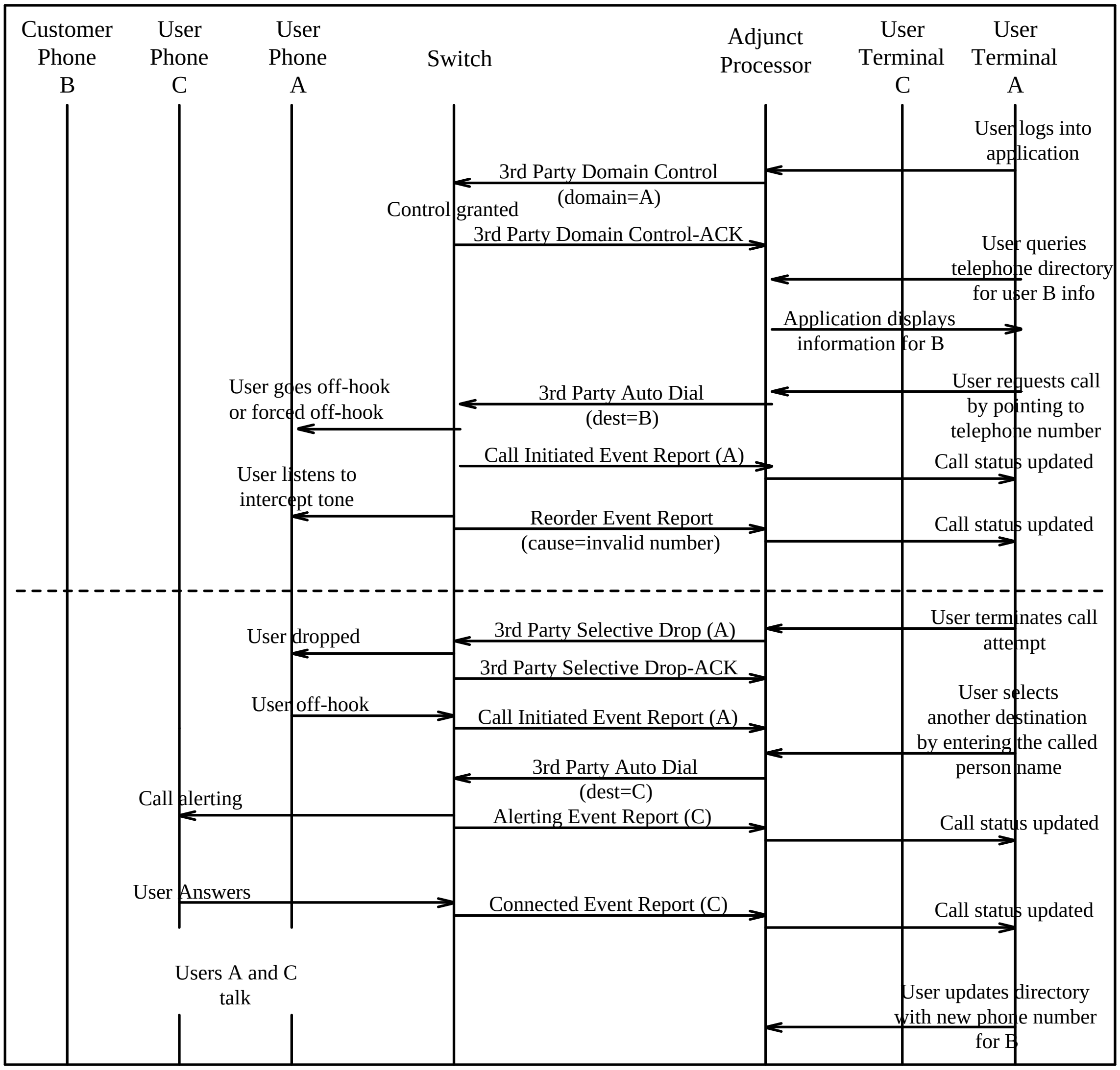
The application in the adjunct processor logs in agents to different splits and changes the work mode of the agents based on the call loads of different ACD splits. The application first checks the status of the ACD splits. Then it moves idle agents from one split to another. In addition, the application changes the work mode of an agent. The application continues to check the status of ACD splits to make further agent moves, if necessary.

Incoming Call Monitoring and Manual Conference — Station Monitoring



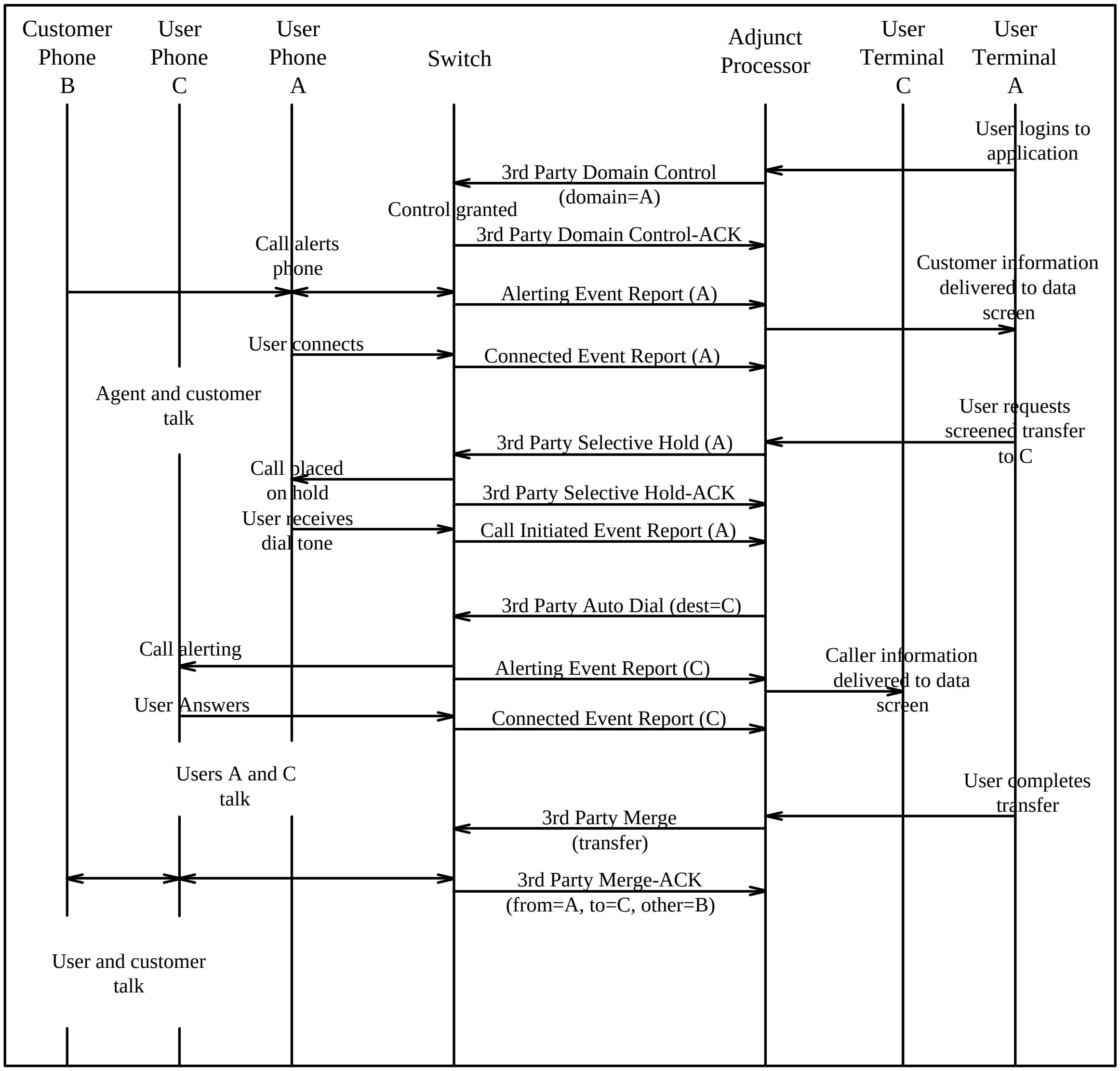
The application in the adjunct processor monitors station A. The station receives a call and conferences another station to the call. The conference operation is done manually at the station by pressing the conference button, dialing the second call, and pressing the conference button a second time.

Screen-Based Dialing — Station Monitoring



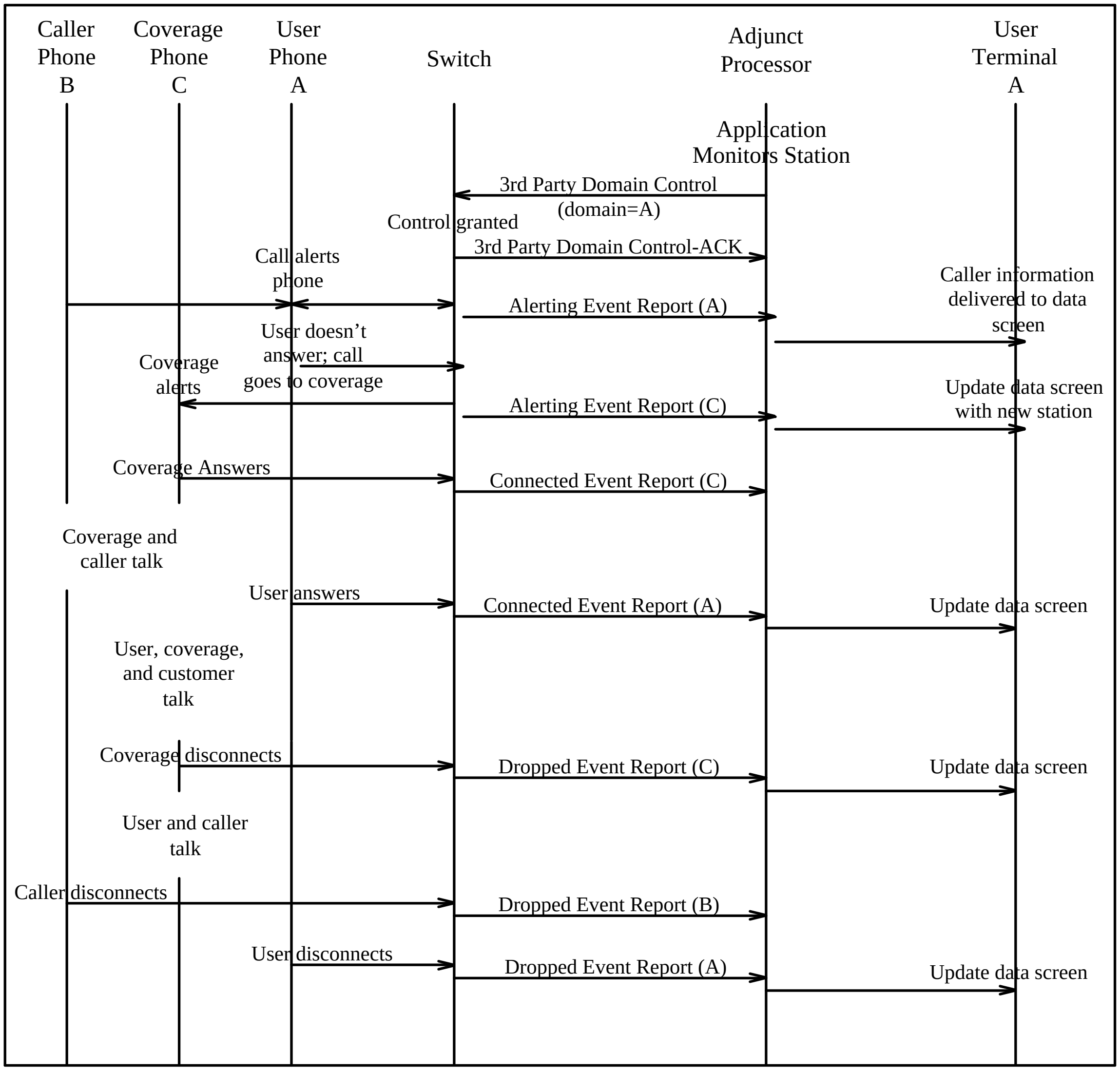
A user at a monitored station initiates a call from the data terminal. The first call receives intercept treatment because the number provided is invalid. The second call is successful.

Screened Transfer — Station Monitoring



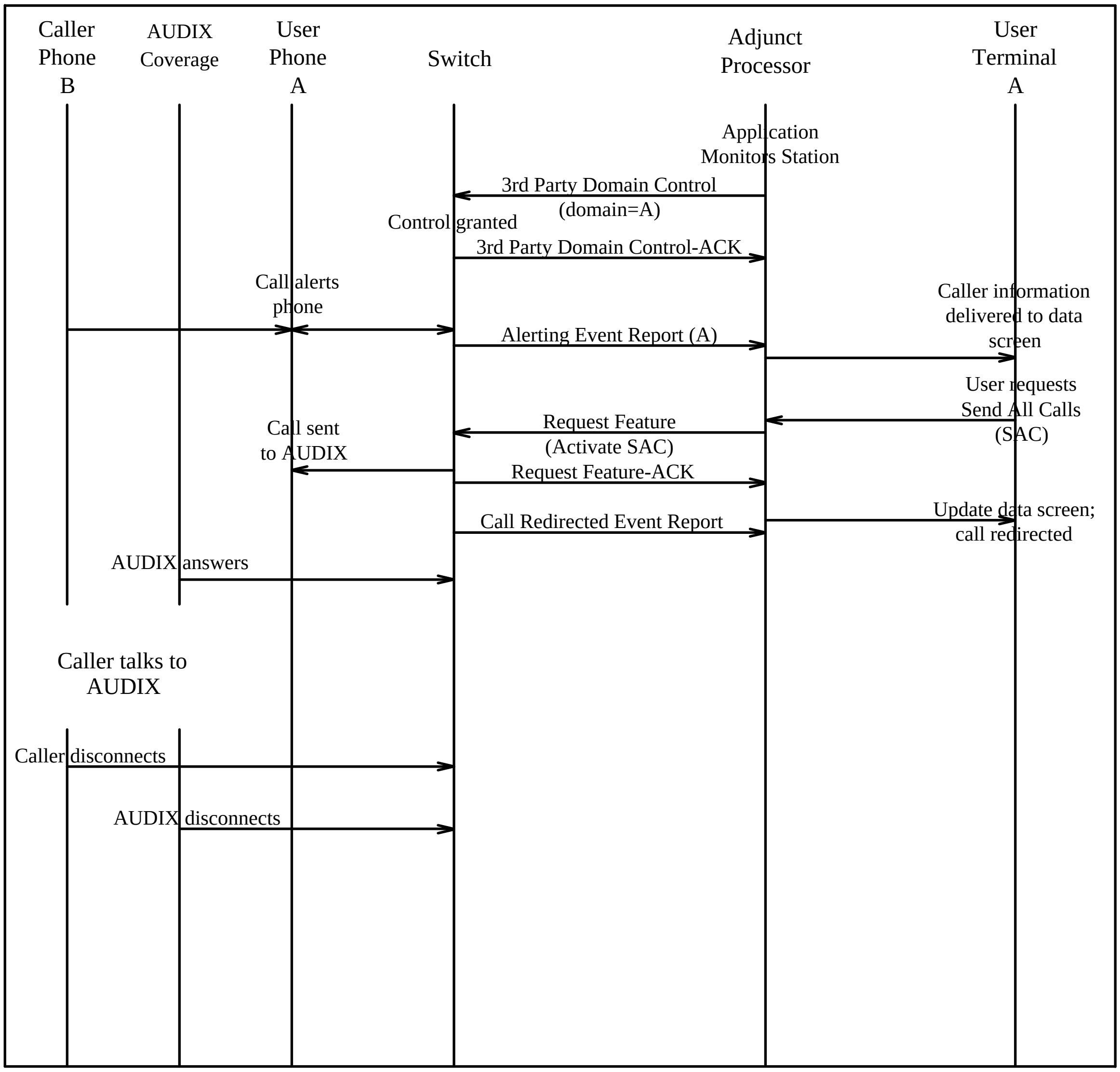
A user at monitored station A performs a screened transfer operation from the data terminal. The adjunct processor, using Third Party Call Control requests, places the first call on hold (**Third Party Selective Hold**) and initiates a call to user C (**Third Party Auto Dial**). After users A and C talk, user A uses the data terminal to signal the adjunct processor to complete the transfer of the held call to user C.

Call Coverage to Station — Station Monitoring



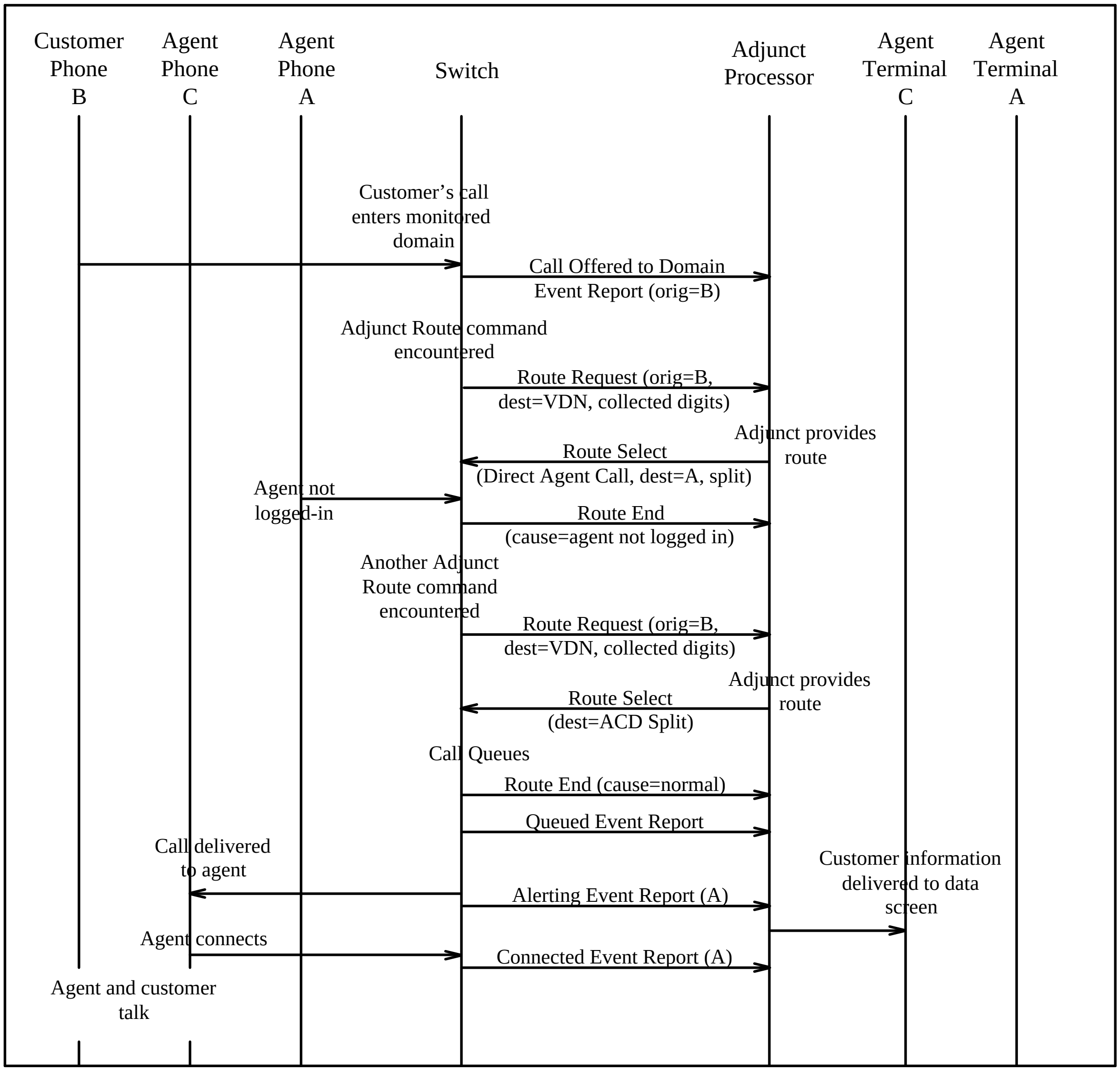
A call to a monitored station goes to coverage after alerting the station. Since the coverage point is another station, the monitored station (original destination) maintains a simulated bridged appearance for the call. The monitored station answers the call after the coverage station has answered the call.

Call Coverage to AUDIX — Station Monitoring



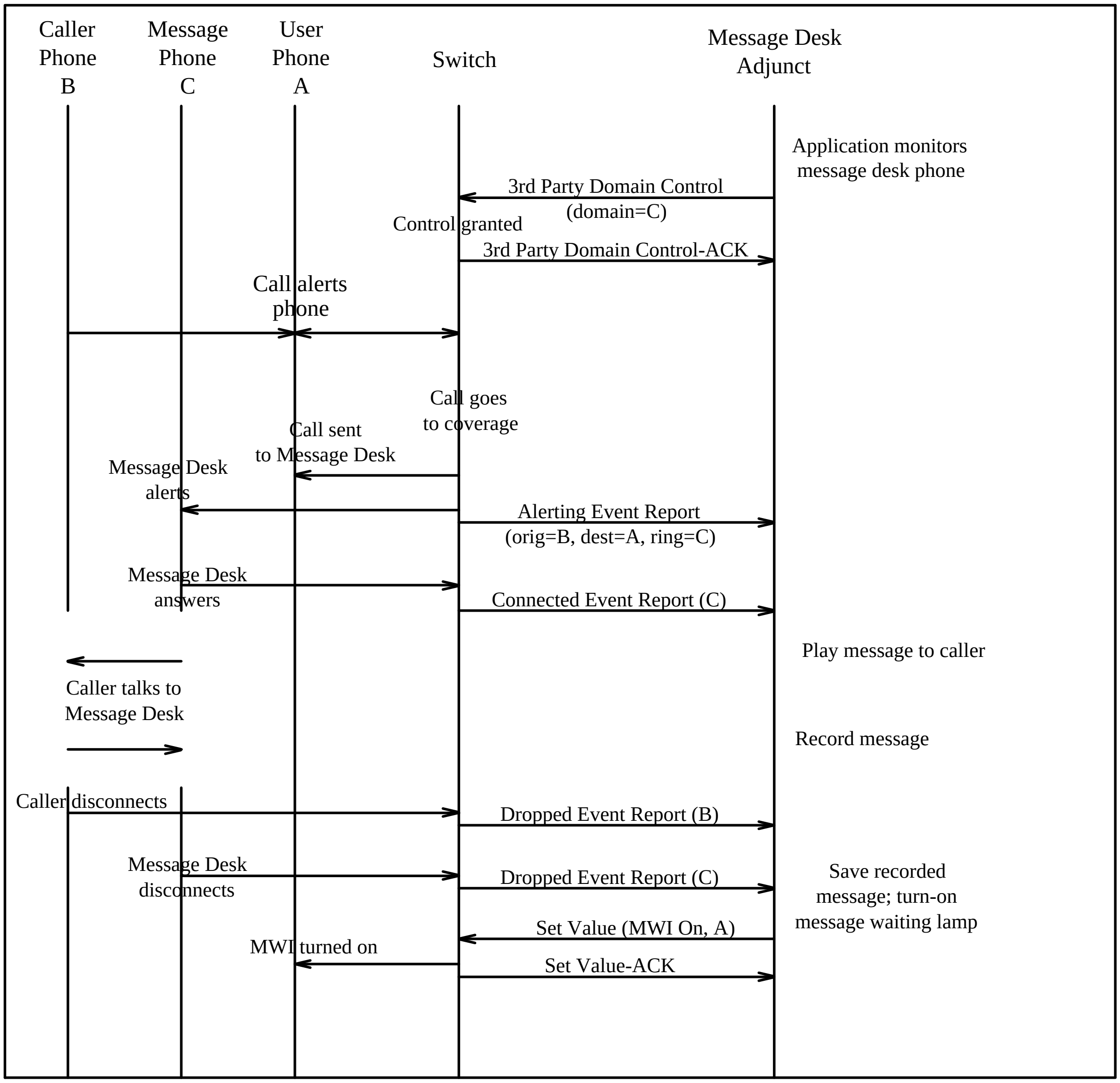
A call to a monitored station goes to AUDIX after alerting the station. Since AUDIX coverage does not maintain a simulated bridged appearance at the monitored station (original destination), the user at the monitored station cannot connect to the call after the call goes to coverage.

Adjunct Routing



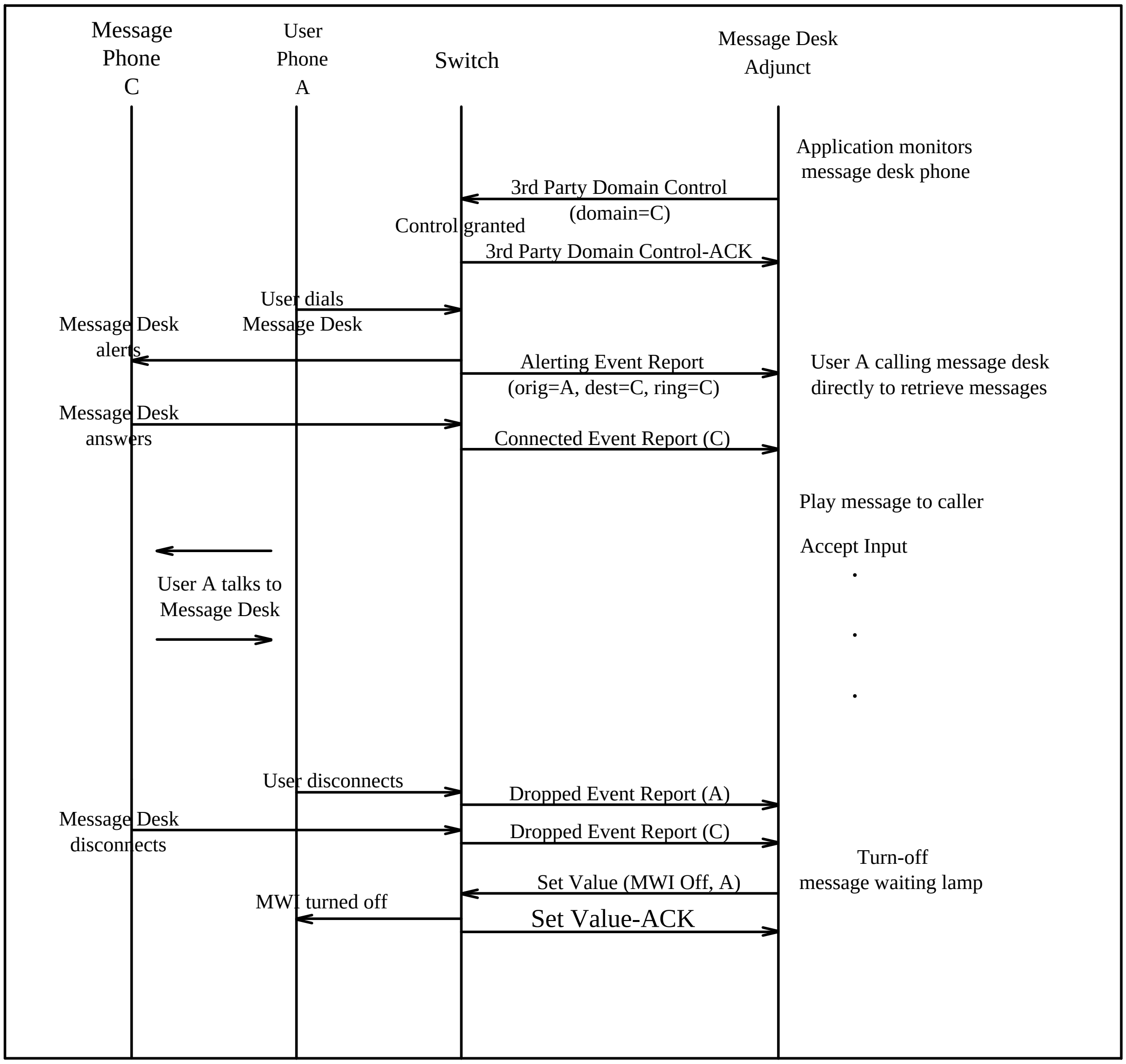
The switch requests a route for a call that encounters an **Adjunct Routing** vector command. The ACD agent at the first destination selected by the application is not logged in. The second destination is an ACD split where the call queues. The application uses the CPN/BN, DNIS, collected digits via Call Prompting, and agent/station availability information to route the call to a local or remote destination.

Message Desk — Incoming Call



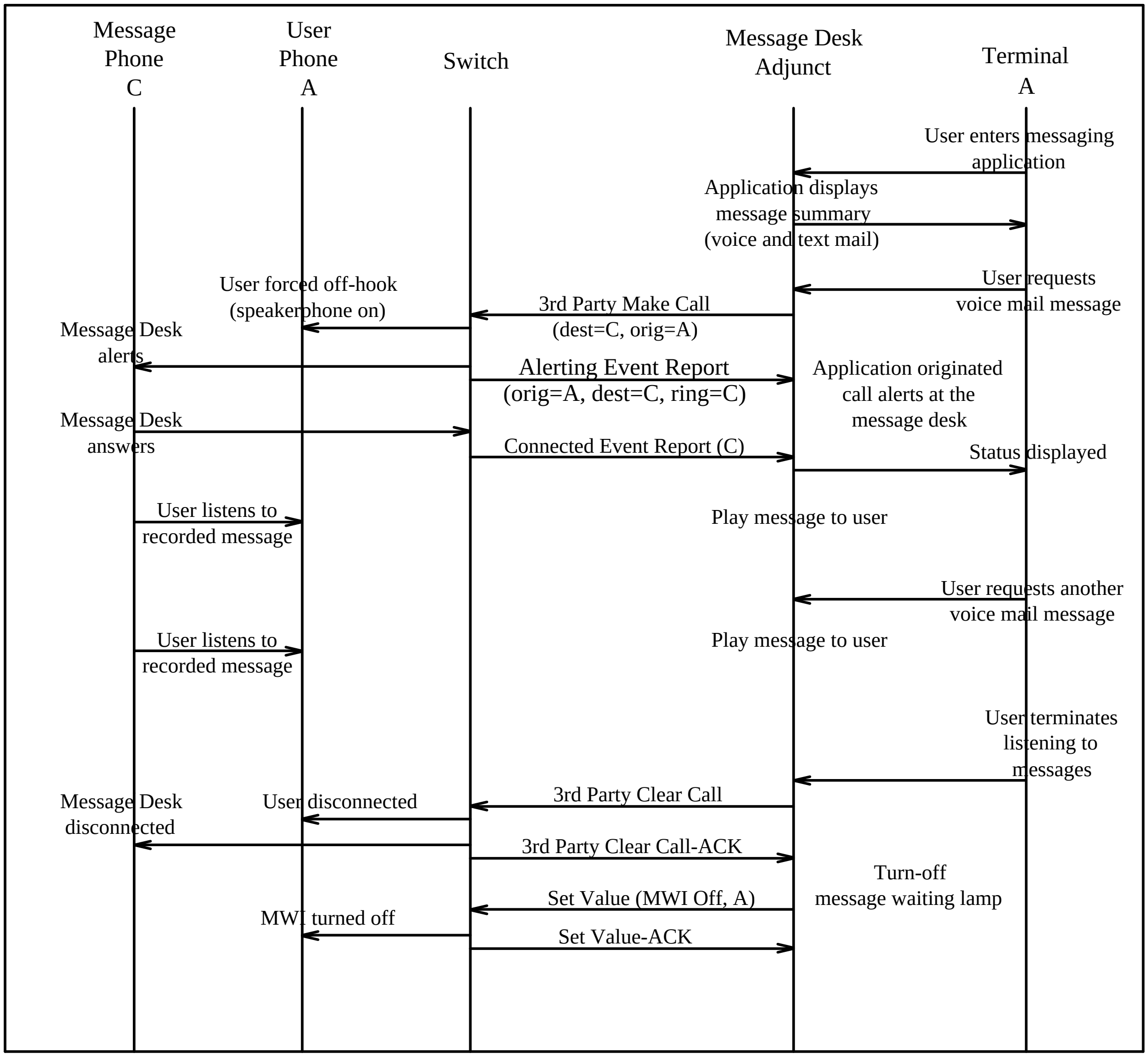
A call to station A redirects to the message desk after alerting the station (original destination). The application only has control of the message desk station (phone).

Message Desk — Message Retrieval via Phone



A user calls the Message Desk to retrieve his or her voice messages. The message desk application knows that the user is requesting his or her messages because the user has dialed the message center directly. The message center application interacts with the user via the voice call.

Message Desk — Message Retrieval via Data Terminal



A user utilizes a data terminal to retrieve his or her messages. The application displays a summary of all voice and text messages stored for the user. The user utilizes the data terminal to interact with the message desk application. When the user wants to listen to a voice mail message, the application forces the user off-hook and originates a call to the message center. The message center then plays the requested message over the phone. The application disconnects the call when no more messages need to be played to the user.

2. Calls Across Multiple Switches

This section presents several scenarios for calls routed, transferred, or conferenced across switches. Figure A-1 shows the VDNs, vectors, splits, and extensions for the following scenarios. Each switch has its own ASAI link, but ASAI links are not shown in the figure.

NOTE:
Section 7 provides examples of calls between switches using UUI.

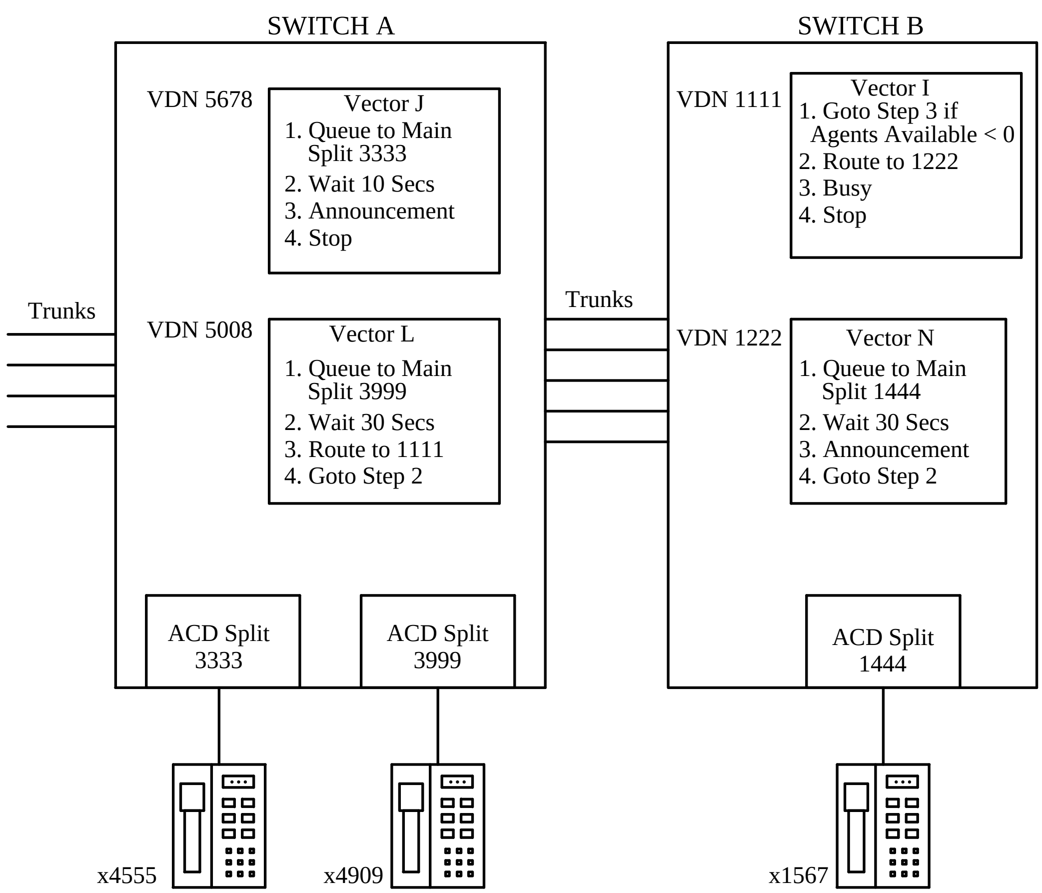


Figure A-1. Multiple Switch Configuration

External Call to VDN, Answered by Station and Transferred to a VDN on Another Switch

This scenario shows the call flow for an incoming ISDN PRI call to VDN 5678 that is answered by extension 4555 in ACD split 3333 (see Figure A-2.) The agent at extension 4555 manually transfers the call to VDN 1222 in switch B. Extension 1567 in ACD split 1444 answers the call at switch B.

The scenario shows the agent at extension 4555 completing the transfer operation while the call is in queue at ACD split 1444. Note that no Alerting or Connected Event Report is sent to switch A, because the call to switch B (call_id 45) is not monitored on switch B until it is merged with the incoming call (call_id 37). If the agent 4555 completes the transfer after talking to agent 1567, the Call Transferred Event Report would have had occurred after the connected Event Report is sent by switch B. All other parameters would have remained the same. Similarly, if the operation is a conference instead of a transfer, the Call Transferred Event Report would have been replaced by a Call Conferenced Event Report.

Assume that VDN 5678 is monitored over CRV 98 by an ASAI Adjunct Processor connected to switch A and that VDN 1222 is monitored over CRV 26 by an ASAI Adjunct Processor connected to switch B. Messages in italics refer to messages exchanged by Switch B and the ASAI Adjunct Processor connected to switch B.

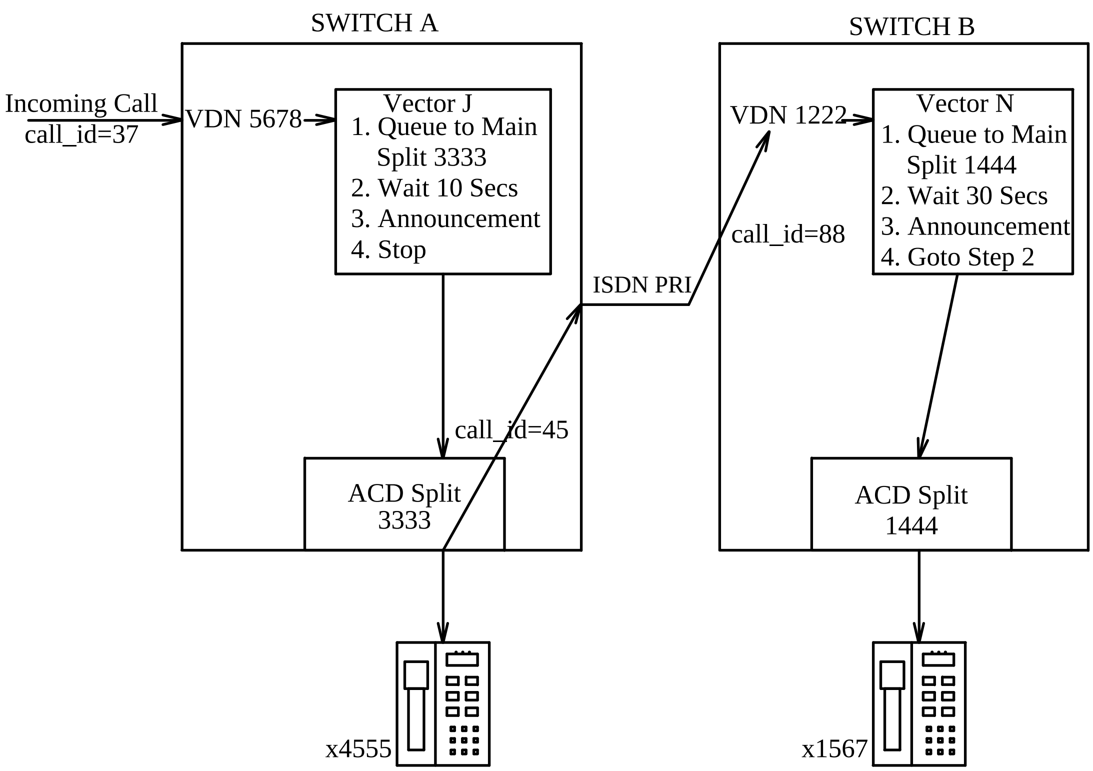
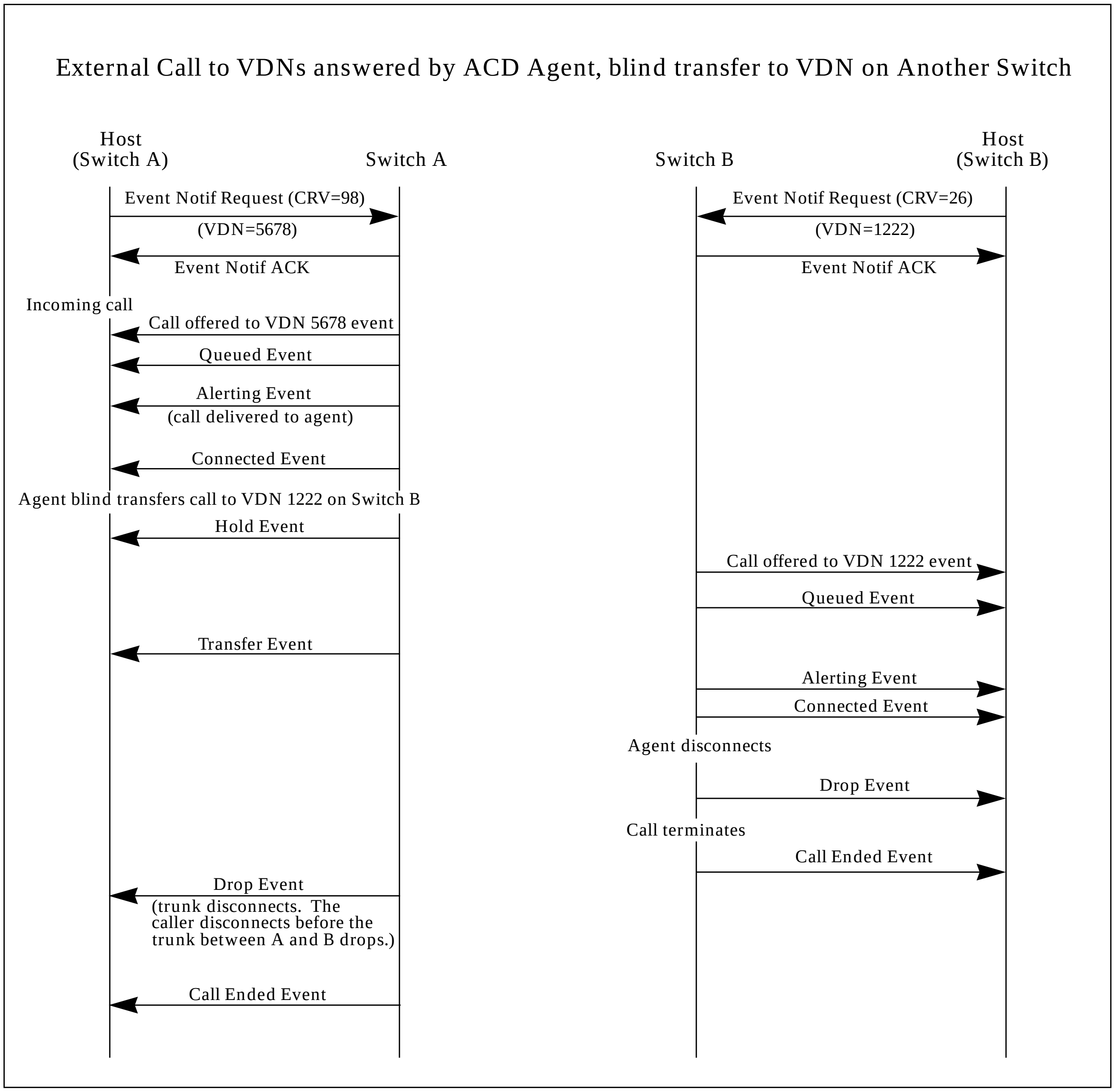


Figure A-2. Call Flow for Blind Transfer to Another Switch


External Call to VDNs answered by ACD Agent, blind transfer to VDN on Another Switch
Host
(Switch A)
Switch A
Switch B
Host
(Switch B)
Event Notif Request (CRV=98)
(VDN=5678)
Event Notif ACK
Event Notif Request (CRV=26)
(VDN=1222)
Event Notif ACK
Incoming call
Call offered to VDN 5678 event
Queued Event
Alerting Event
(call delivered to agent)
Connected Event
Agent blind transfers call to VDN 1222 on Switch B
Hold Event
Call offered to VDN 1222 event
Queued Event
Transfer Event
Alerting Event
Connected Event
Agent disconnects
Drop Event
Call terminates
Call Ended Event
Drop Event
(trunk disconnects. The
caller disconnects before the
trunk between A and B drops.)
Call Ended Event

External Call to VDN, Answered by Station, and Transferred to a Station on Another Switch

This scenario shows the call flow for an incoming non-ISDN call to VDN 5678 that is answered by extension 4555 in ACD split 3333 (see Figure A-3). The agent at extension 4555 does a consultation transfer to extension 1567 on switch B. That is, the transfer is completed after the agent on extension 4555 talks to the agent on extension 1567. The trunks between switches are ISDN PRI trunks.

Assume that VDN 5678 is monitored over CRV 98 by an ASAI Adjunct Processor connected to switch A and that extension 1567 is domain-controlled over CRV 45 by an ASAI Adjunct Processor connected to switch B. Messages in italics refer to messages exchanged by switch B and the ASAI Adjunct Processor connected to switch B.

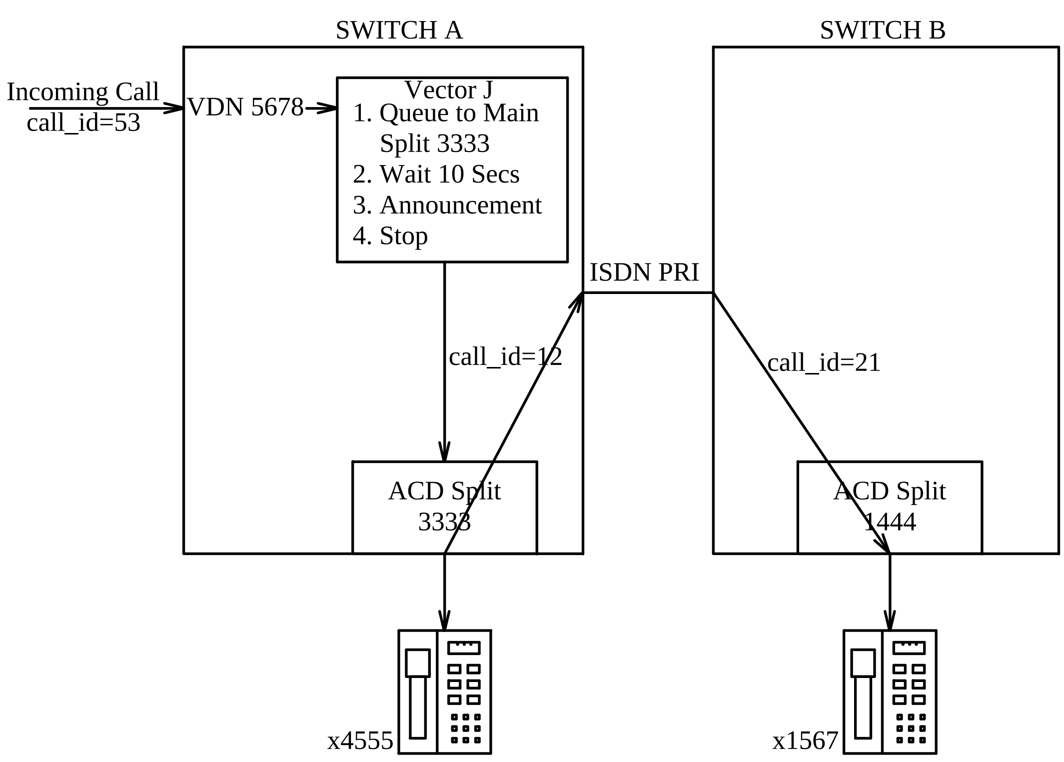
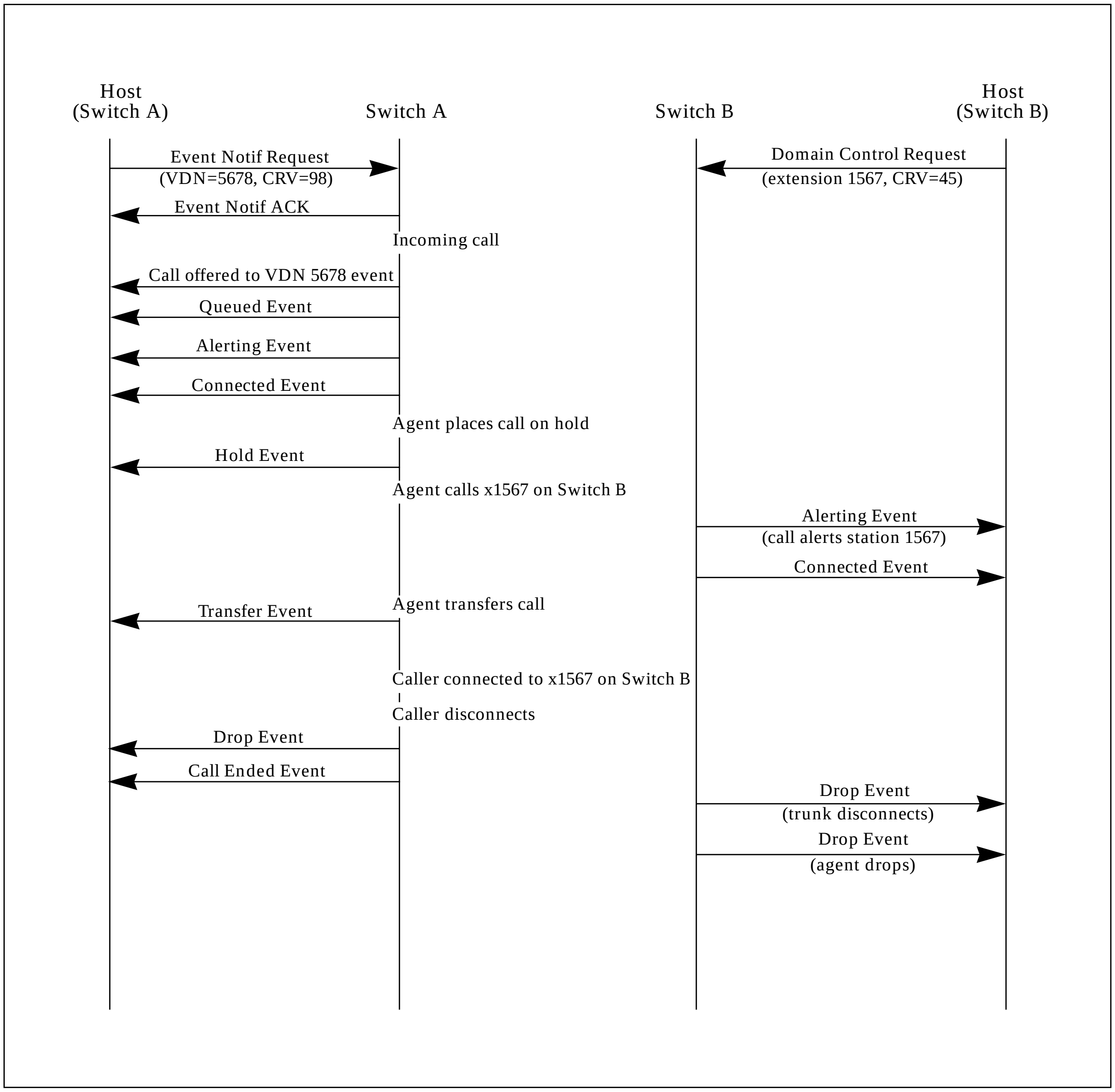


Figure A-3. Call Flow for Consultation Transfer to Another Switch


Host
(Switch A)
Switch A
Switch B
Host
(Switch B)
Event Notif Request
(VDN=5678, CRV=98)
Domain Control Request
(extension 1567, CRV=45)
Event Notif ACK
Incoming call
Call offered to VDN 5678 event
Queued Event
Alerting Event
Connected Event
Agent places call on hold
Hold Event
Agent calls x1567 on Switch B
Alerting Event
(call alerts station 1567)
Connected Event
Transfer Event
Agent transfers call
Caller connected to x1567 on Switch B
Caller disconnects
Drop Event
Call Ended Event
Drop Event
(trunk disconnects)
Drop Event
(agent drops)

External Call to Lookahead Interflow VDN

This scenario shows the call flow for an incoming ISDN PRI call to VDN 5008 that looks ahead to VDN 1111 on switch B (see Figure A-4). The first lookahead interflow attempt is denied by switch B. The second lookahead interflow attempt is accepted by switch B and the call is delivered to and answered by extension 1567 in ACD split 1444.

Note that switch A is not guaranteed to receive an ISDN Alerting message from switch B. In these cases, the ASAI adjunct connected to switch A does not receive the Alerting Event Report. The lookahead display information is the VDN name provided on the VDN administration form and is subject to the VDN display override rules.

Assume that VDN 5008 is monitored over CRV 80 by an ASAI Adjunct Processor connected to switch A, and that VDNs 1111 and 1222 are monitored over CRVs 20 and 26, respectively, by an ASAI Adjunct Processor connected to switch B. Messages in italics refer to messages exchanged by Switch B and the ASAI Adjunct Processor connected to switch B.

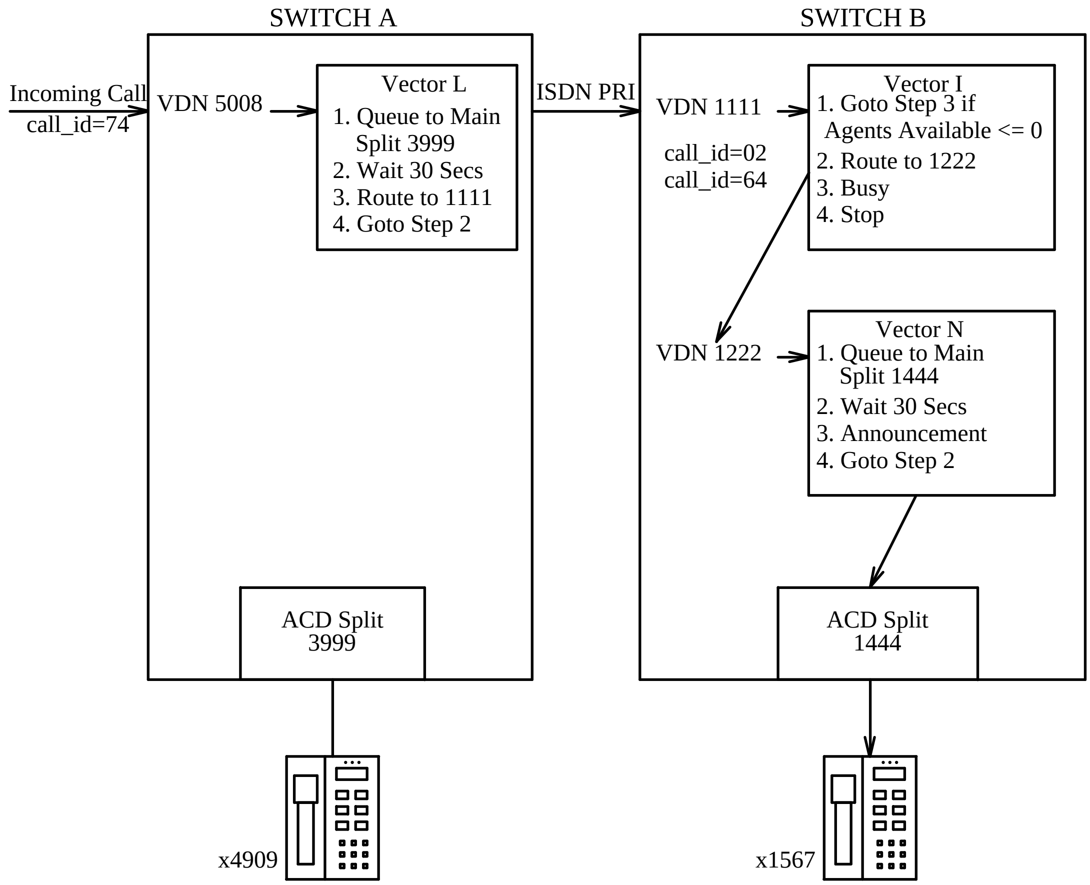
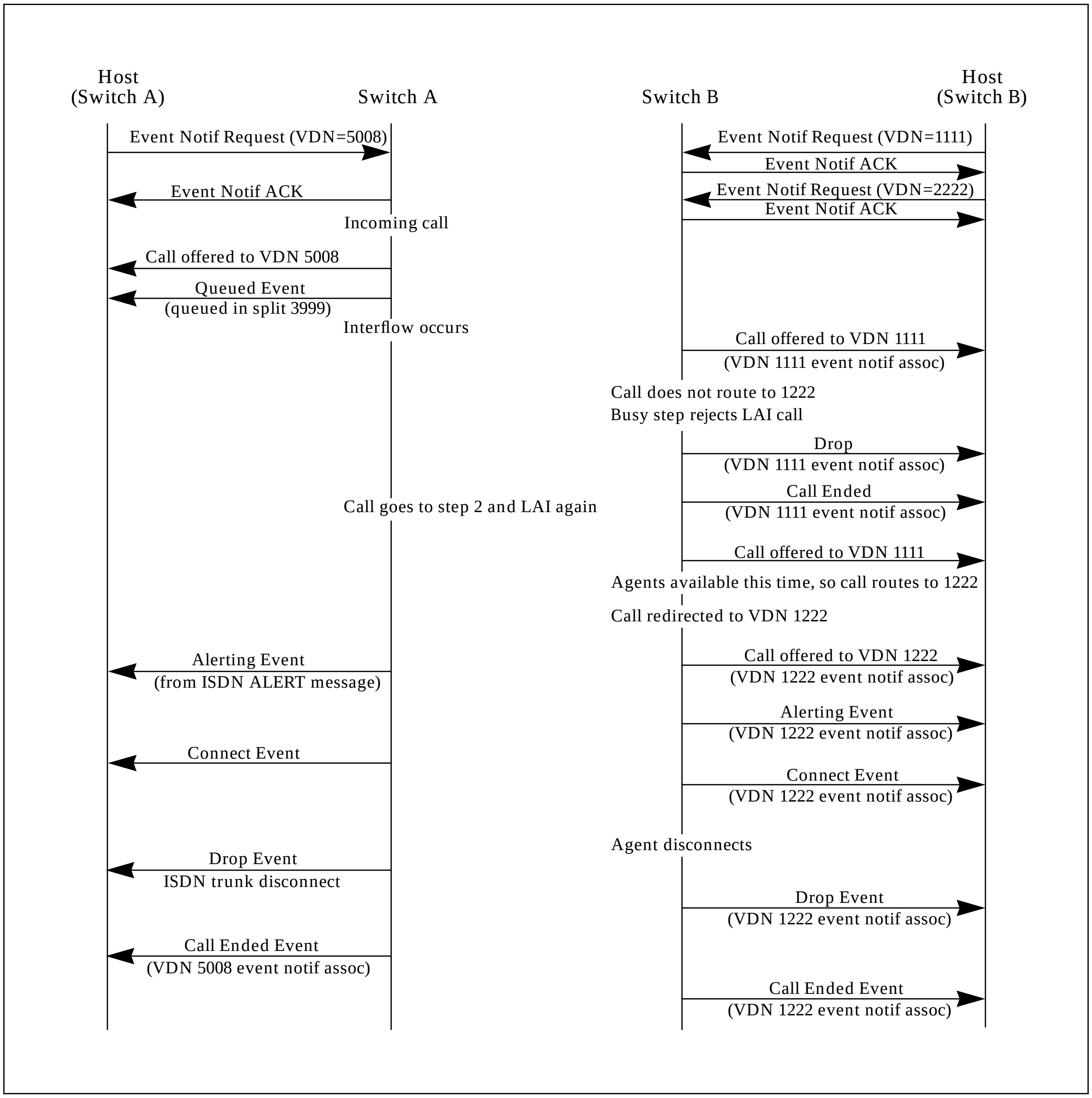


Figure A-4. Call Flow for Incoming Call to Lookahead Interflow Vector


Host
(Switch A)
Switch A
Switch B
Host
(Switch B)
Event Notif Request (VDN=5008)
Event Notif Request (VDN=1111)
Event Notif ACK
Event Notif ACK
Event Notif Request (VDN=2222)
Event Notif ACK
Incoming call
Call offered to VDN 5008
Queued Event
(queued in split 3999)
Interflow occurs
Call offered to VDN 1111
(VDN 1111 event notif assoc)
Call does not route to 1222
Busy step rejects LAI call
Drop
(VDN 1111 event notif assoc)
Call Ended
(VDN 1111 event notif assoc)
Call goes to step 2 and LAI again
Call offered to VDN 1111
Agents available this time, so call routes to 1222
Call redirected to VDN 1222
Alerting Event
(from ISDN ALERT message)
Call offered to VDN 1222
(VDN 1222 event notif assoc)
Alerting Event
(VDN 1222 event notif assoc)
Connect Event
Connect Event
(VDN 1222 event notif assoc)
Agent disconnects
Drop Event
ISDN trunk disconnect
Drop Event
(VDN 1222 event notif assoc)
Call Ended Event
(VDN 5008 event notif assoc)
Call Ended Event
(VDN 1222 event notif assoc)

External Call to VDN, Answered by a Local Station, and Transferred to a Lookahead Interflow VDN

This scenario shows the call flow for an incoming ISDN PRI call to VDN 5678 that is answered by extension 4555 in ACD split 3333 and subsequently transferred to lookahead interflow VDN 5008 (see Figure A-5). The call is accepted by the receiving switch (switch B), waits in queue until the vector announcement (VDN 1222, vector N) and is abandoned by the caller while the call is alerting extension 1567. The lookahead display information is the VDN name provided on the VDN administration form.

The transfer to VDN 5008 is completed before the call attempts the lookahead to switch B. If the call had initiated the lookahead interflow before the transfer operation had been completed, the ASAI adjunct connected to switch B would have received the extension 4555 as the calling party number instead of the original SID/ANI for the call.

Note that the Alerting Event Report received by the ASAI Adjunct Processor connected to switch A is received only if the lookahead interflowed call receives alerting treatment (that is, wait hearing ringback/announcement or extension alerting) before the call is answered on switch B. If the call waits with silence (for example, wait hearing silence) before switch B applies ringback or answers the call, the ASAI Adjunct Processor connected to switch A receives a Cut-Through/Progress Event Report. Subsequent Alerting and Connected Event Reports are provided, depending on the call treatment provided by vector processing.

The Connected Event Report received by the ASAI Adjunct Processor connected to switch A is triggered by the ISDN Connect message received from switch B. Switch B provides a single ISDN Connect message with the first answer treatment provided. For example, listening to music or to an announcement while the call is in vector processing triggers switch B to send an ISDN Connect message to switch A. Switch B does not send additional ISDN messages to switch A for subsequent answers (for example, other announcements or answered by a station). Therefore, the ASAI Adjunct Processor connected to switch A does not receive further answer/connect notifications for the call.

Also note that the ISDN called number received by the adjunct connected to switch A is not the same as the VDN number, since ISDN digit manipulation has occurred.

Assume that VDNs 5678 and 5008 are monitored over CRVs 98 and 80, respectively, by an ASAI Adjunct Processor connected to switch A and that VDNs 1111 and 1222 are monitored over CRVs 20 and 26, respectively, by an ASAI Adjunct Processor connected to switch B. Messages in italics refer to messages exchanged by Switch B and the ASAI Adjunct Processor connected to switch B.

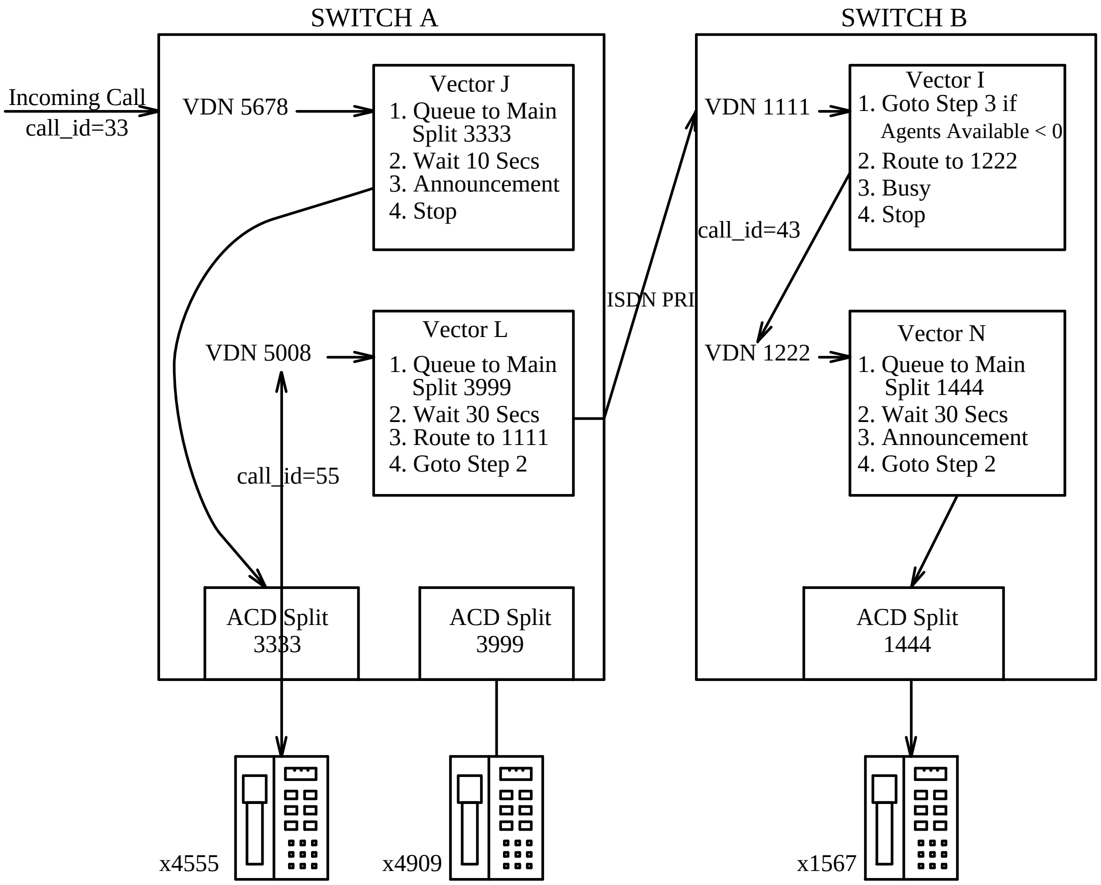
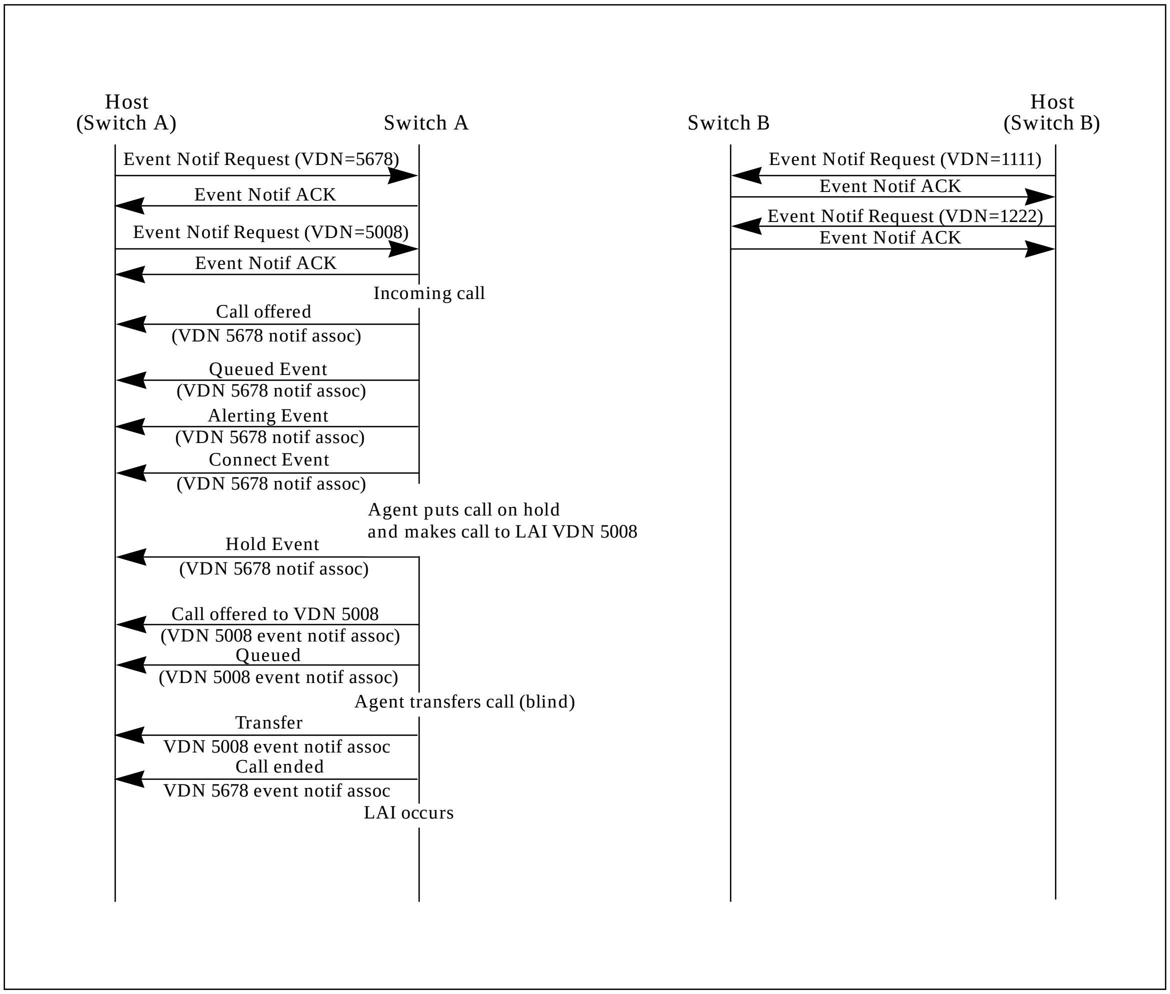
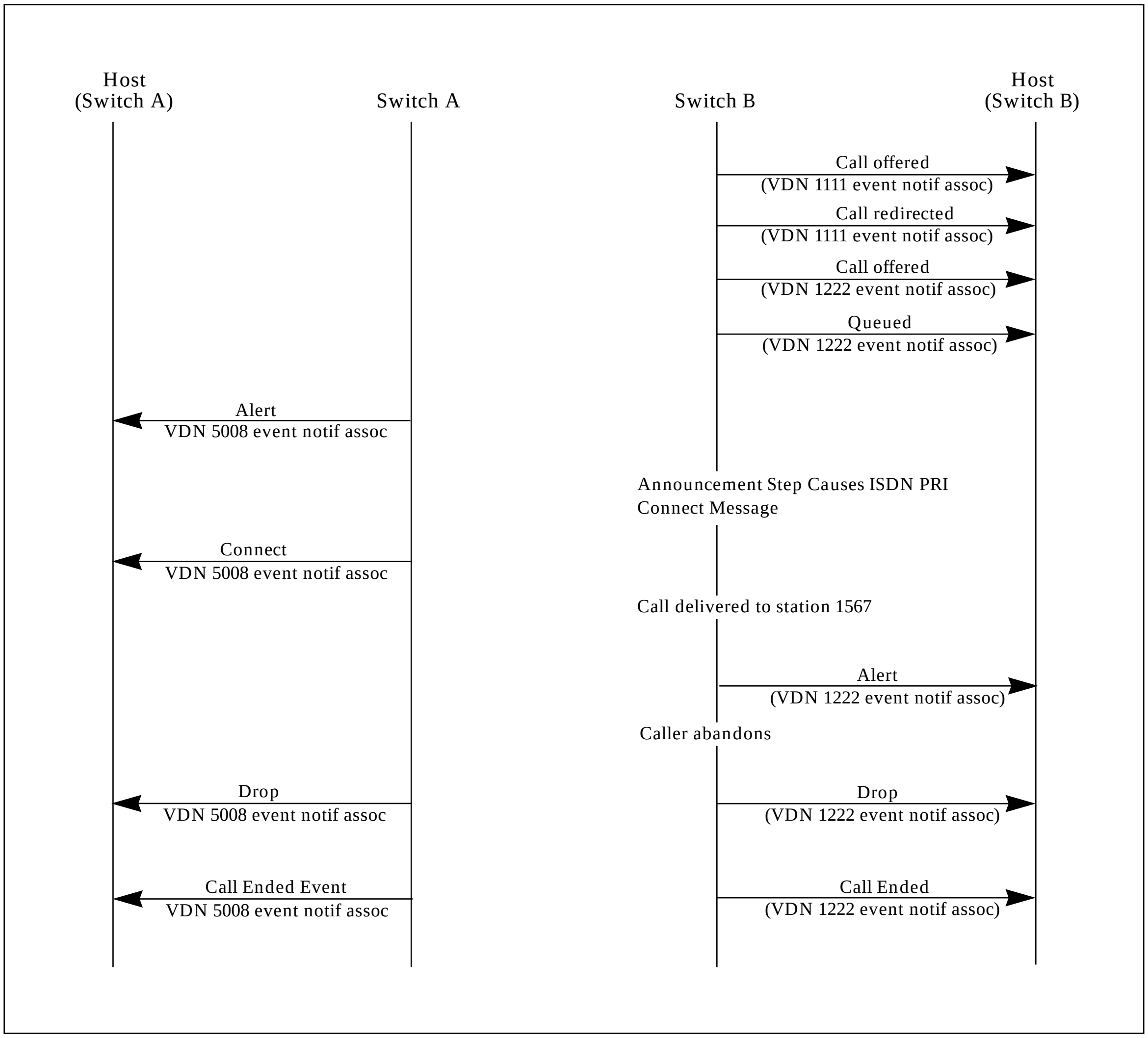


Figure A-5. Call Flow for a Transfer to a Lookahead Interflow Vector


Host
(Switch A)
Switch A
Switch B
Host
(Switch B)
Event Notif Request (VDN=5678)
Event Notif ACK
Event Notif Request (VDN=5008)
Event Notif ACK
Event Notif Request (VDN=1111)
Event Notif ACK
Event Notif Request (VDN=1222)
Event Notif ACK
Incoming call
Call offered
(VDN 5678 notif assoc)
Queued Event
(VDN 5678 notif assoc)
Alerting Event
(VDN 5678 notif assoc)
Connect Event
(VDN 5678 notif assoc)
Agent puts call on hold
and makes call to LAI VDN 5008
Hold Event
(VDN 5678 notif assoc)
Call offered to VDN 5008
(VDN 5008 event notif assoc)
Queued
(VDN 5008 event notif assoc)
Agent transfers call (blind)
Transfer
VDN 5008 event notif assoc
Call ended
VDN 5678 event notif assoc
LAI occurs


Host
(Switch A)
Switch A
Switch B
Host
(Switch B)
Call offered
(VDN 1111 event notif assoc)
Call redirected
(VDN 1111 event notif assoc)
Call offered
(VDN 1222 event notif assoc)
Queued
(VDN 1222 event notif assoc)
Alert
VDN 5008 event notif assoc
Announcement Step Causes ISDN PRI
Connect Message
Connect
VDN 5008 event notif assoc
Call delivered to station 1567
Alert
(VDN 1222 event notif assoc)
Caller abandons
Drop
VDN 5008 event notif assoc
Drop
(VDN 1222 event notif assoc)
Call Ended Event
VDN 5008 event notif assoc
Call Ended
(VDN 1222 event notif assoc)

3. Expert Agent Selection Interactions

This section presents call scenarios in the Expert Agent Selection Environment.

External Call to VDN, Answered by Logical Agent, and Conferenced with Another Logical Agent

This scenario shows an incoming non-ISDN call to VDN 5555 that queues to skills 3333 and 4444 (see Figure A-6). Logical agent 2345, logged in from station 6666, answers the call and conferences logical agent 8766 (logged in from station 9999). No queue event is provided for skill 4444 because logical agent 2345 with skill 4444 is available immediately to answer the call.

Note that the **called number** in the Call Conference Event Report provides the agent's physical extension, not the agent's login id extension.

Event Notification for VDN 5555 is active over CRV 78 as shown at the beginning of the call flow. The Adjunct Processor-initiated login for logical agent 2345 is also shown. Logical Agent 8766 is assumed to have logged in manually at the voice station.

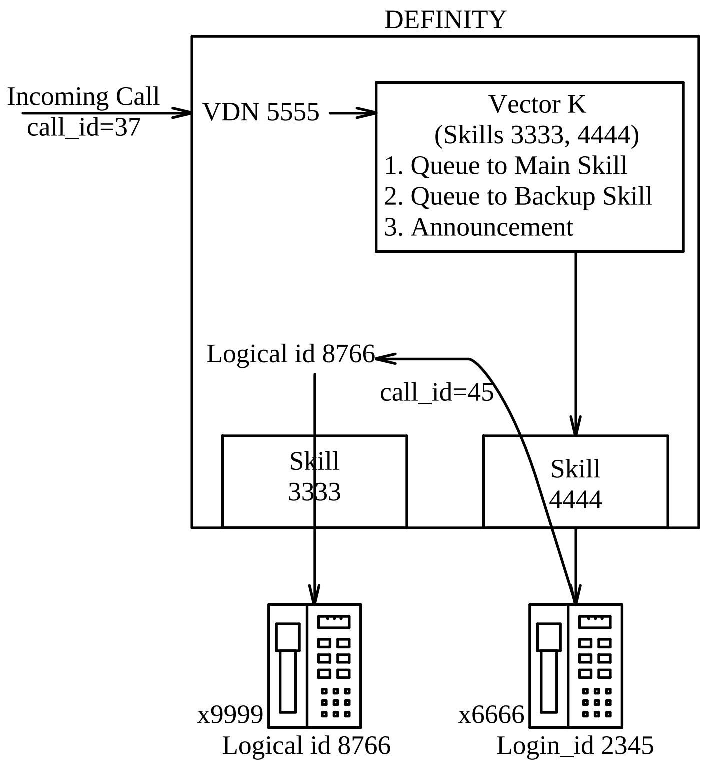
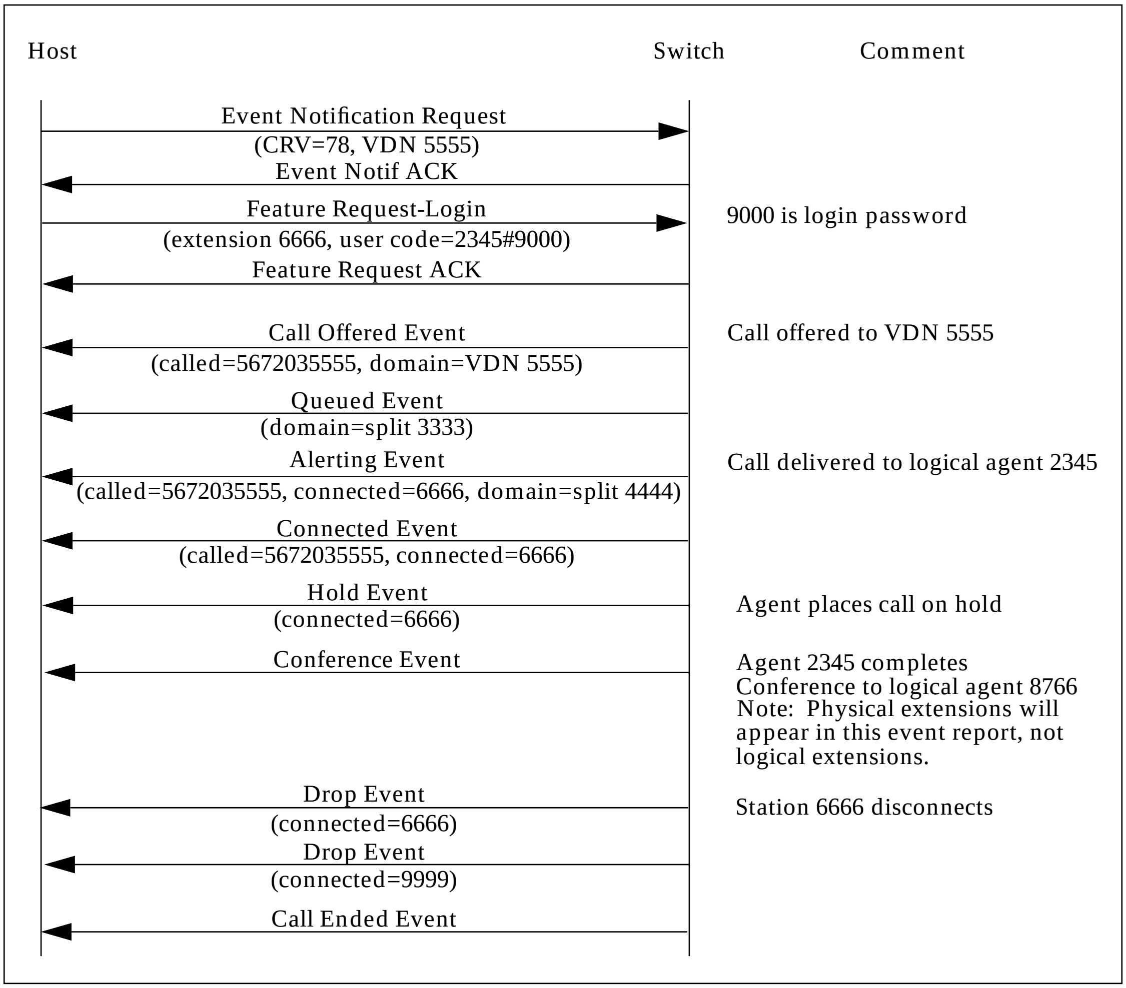


Figure A-6. Call Flow for Incoming Call to Skill VDN


Host
Switch
Comment
Event Notification Request
(CRV=78, VDN 5555)
Event Notif ACK
Feature Request-Login
(extension 6666, user code=2345#9000)
9000 is login password
Feature Request ACK
Call Offered Event
(called=5672035555, domain=VDN 5555)
Call offered to VDN 5555
Queued Event
(domain=split 3333)
Alerting Event
(called=5672035555, connected=6666, domain=split 4444)
Call delivered to logical agent 2345
Connected Event
(called=5672035555, connected=6666)
Hold Event
(connected=6666)
Agent places call on hold
Conference Event
Agent 2345 completes
Conference to logical agent 8766
Note: Physical extensions will
appear in this event report, not
logical extensions.
Drop Event
(connected=6666)
Station 6666 disconnects
Drop Event
(connected=9999)
Call Ended Event

External Call to a Logical Agent´s Station Transferred to Another Logical Agent

This scenario shows an incoming ISDN PRI call to a domain-controlled station, 6666 (see Figure A-7). Logical agent 2345 (logged in from station 6666) answers the call and transfers it to logical agent 8766 logged in from station 9999. Logical agent 3456 completes the transfer operation while the call is queued for logical agent 8766.

Note that the **called number** in the Transfer Event Report contains the logical agent's login id extension. If the transferred operation had occurred after the call was delivered to an agent station, the **called party** would have contained the physical station's extension. A call is delivered to a station if the call is either alerting or connected to the station.

Domain Controls for stations 6666 and 9999 are active over CRV 56 and 34, respectively, as shown at the beginning of the call flow. Third Party Domain Control is only allowed on a physical extension number; it is not allowed on a login id extension.

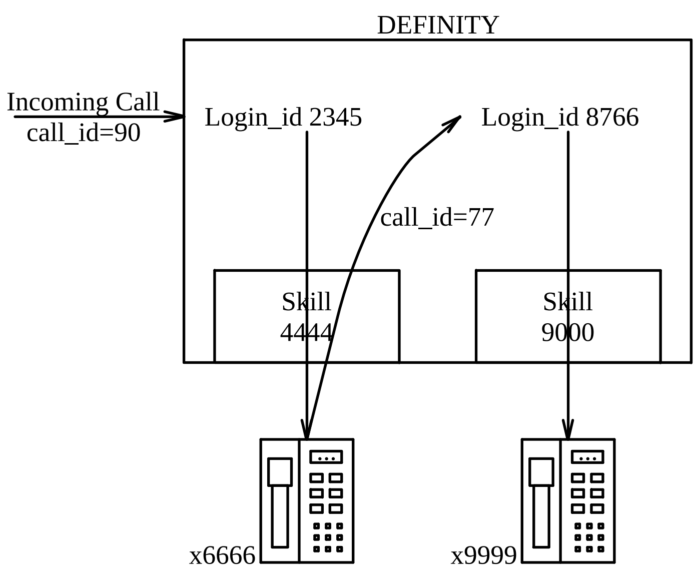
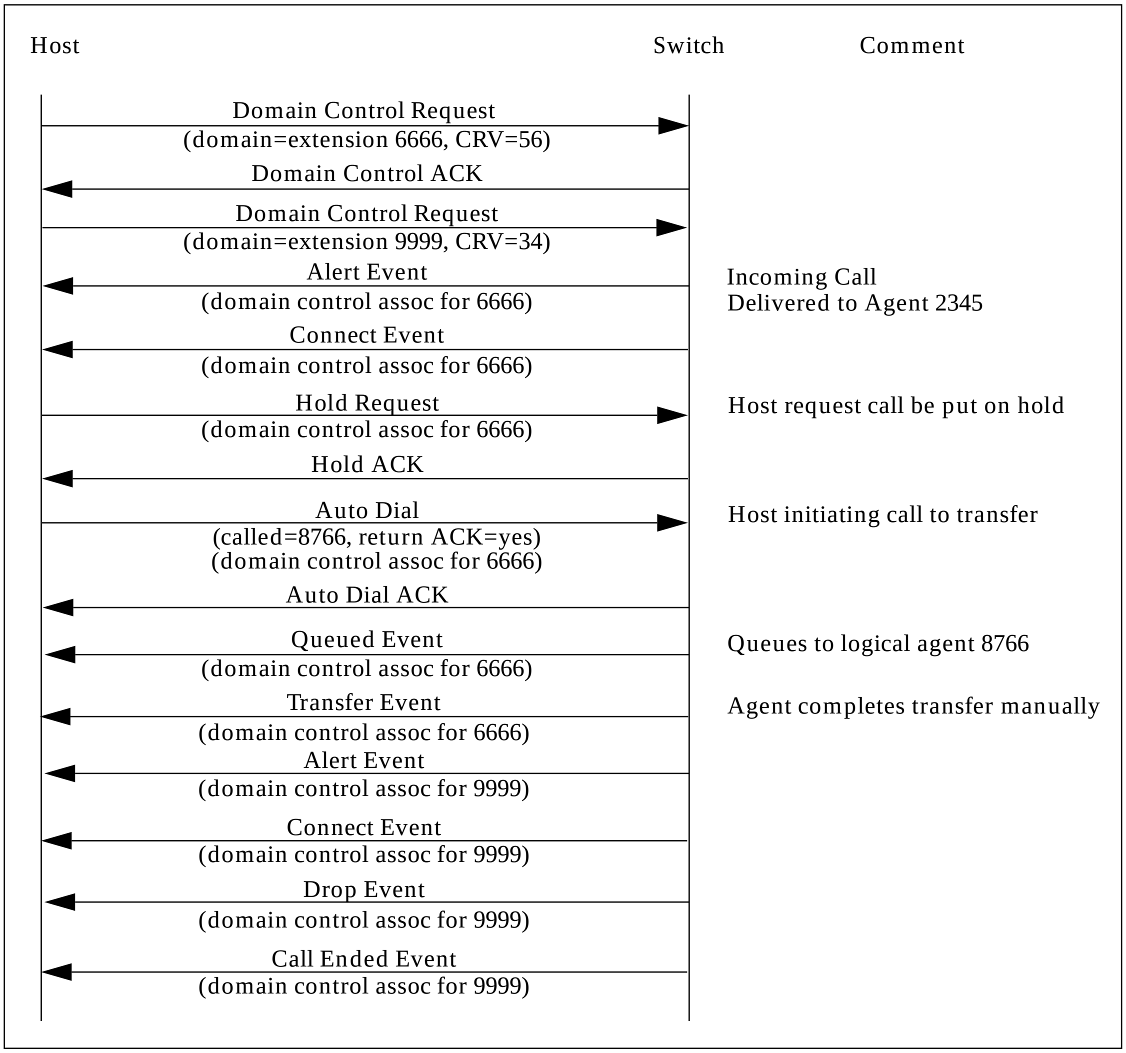


Figure A-7. Call Flow for Incoming Call to Logical Agent Transferred to Another Logical Agent


Host
Switch
Comment
Domain Control Request
(domain=extension 6666, CRV=56)
Domain Control ACK
Domain Control Request
(domain=extension 9999, CRV=34)
Alert Event
(domain control assoc for 6666)
Incoming Call
Delivered to Agent 2345
Connect Event
(domain control assoc for 6666)
Hold Request
(domain control assoc for 6666)
Host request call be put on hold
Hold ACK
Auto Dial
(called=8766, return ACK=yes)
(domain control assoc for 6666)
Host initiating call to transfer
Auto Dial ACK
Queued Event
(domain control assoc for 6666)
Queues to logical agent 8766
Transfer Event
(domain control assoc for 6666)
Agent completes transfer manually
Alert Event
(domain control assoc for 9999)
Connect Event
(domain control assoc for 9999)
Drop Event
(domain control assoc for 9999)
Call Ended Event
(domain control assoc for 9999)

Direct Agent Call to Logical Agent — Make Call to Login ID

This scenario shows the call flow for a **Third Party Make Call** from logical agent 2345 to logical agent 8766 (see Figure A-8). Logical agent 2345 is logged in from station 6666, and logical agent 8766 is logged in from station 9999. Logical agent 8766 is not available to receive the call and the call goes to the coverage destination for login id 8766 (as opposed to following the coverage path associated with station 8900).

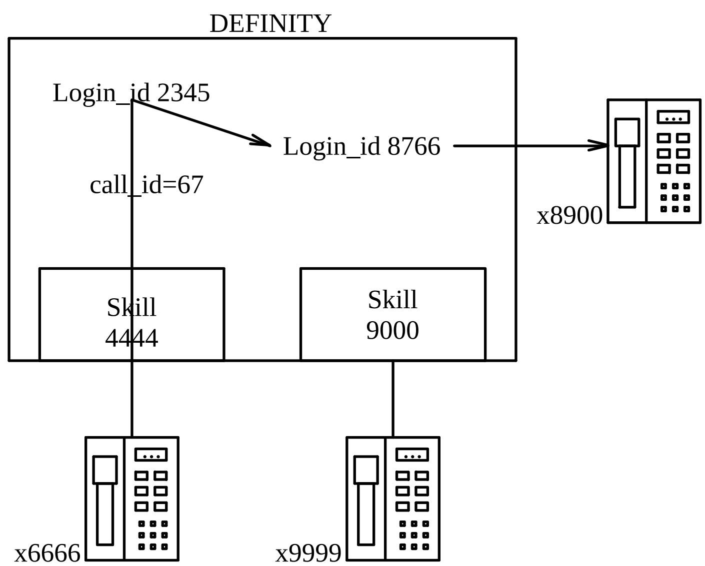
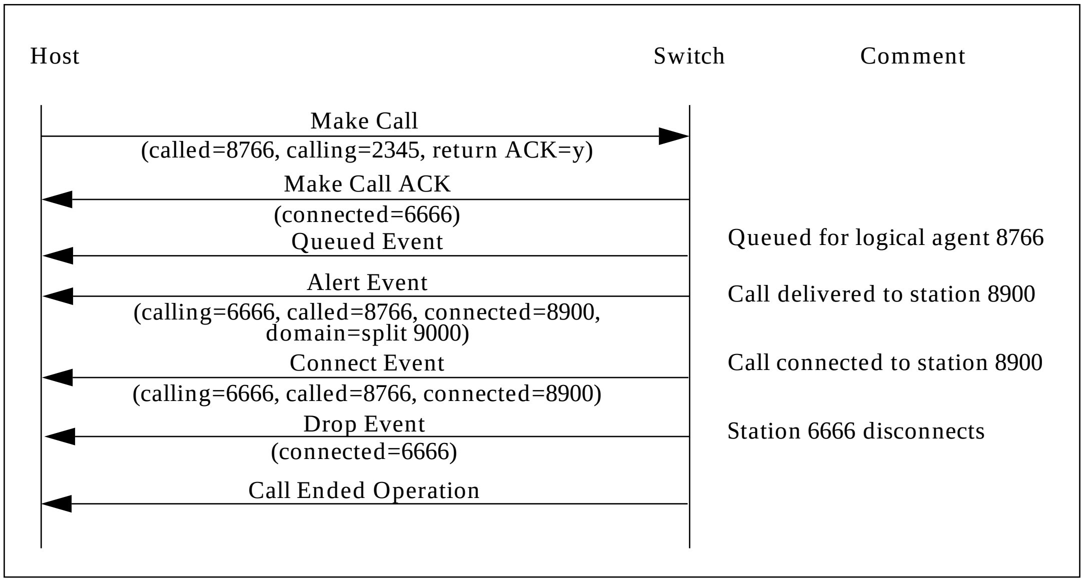


Figure A-8. Call Flow for Direct Agent Call to Logical Agent's Login ID


Host
Switch
Comment
Make Call
(called=8766, calling=2345, return ACK=y)
Make Call ACK
(connected=6666)
Queued Event
Queued for logical agent 8766
Alert Event
Call delivered to station 8900
(calling=6666, called=8766, connected=8900,
domain=split 9000)
Connect Event
Call connected to station 8900
(calling=6666, called=8766, connected=8900)
Drop Event
Station 6666 disconnects
(connected=6666)
Call Ended Operation

Value Queries for Logical Agent and Skill Hunt Groups

This scenario shows the Login Audit Query, ACD Agent Status Query, and Extension Query for skill hunt group 4444 and logical agents 2345, 8766, and 6777 (see Figure A-9). Logical agents 2345 and 8766 are logged into skill 4444 from stations 6666 and 9999, respectively. Logical agent 6777 is not logged in.

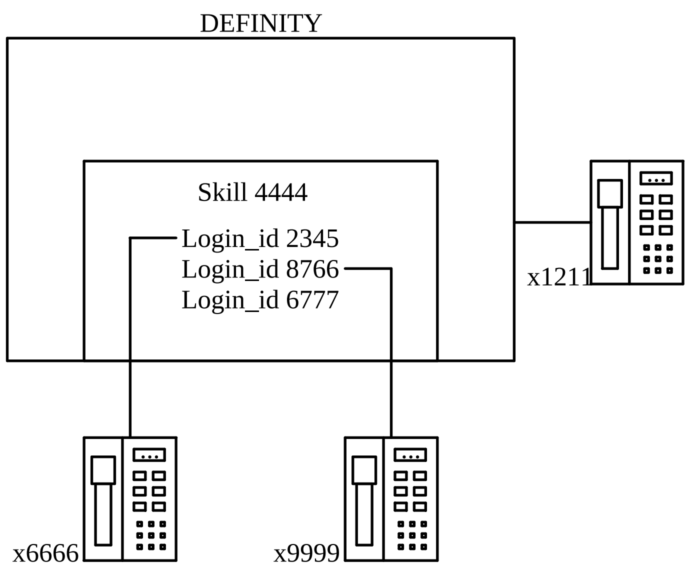
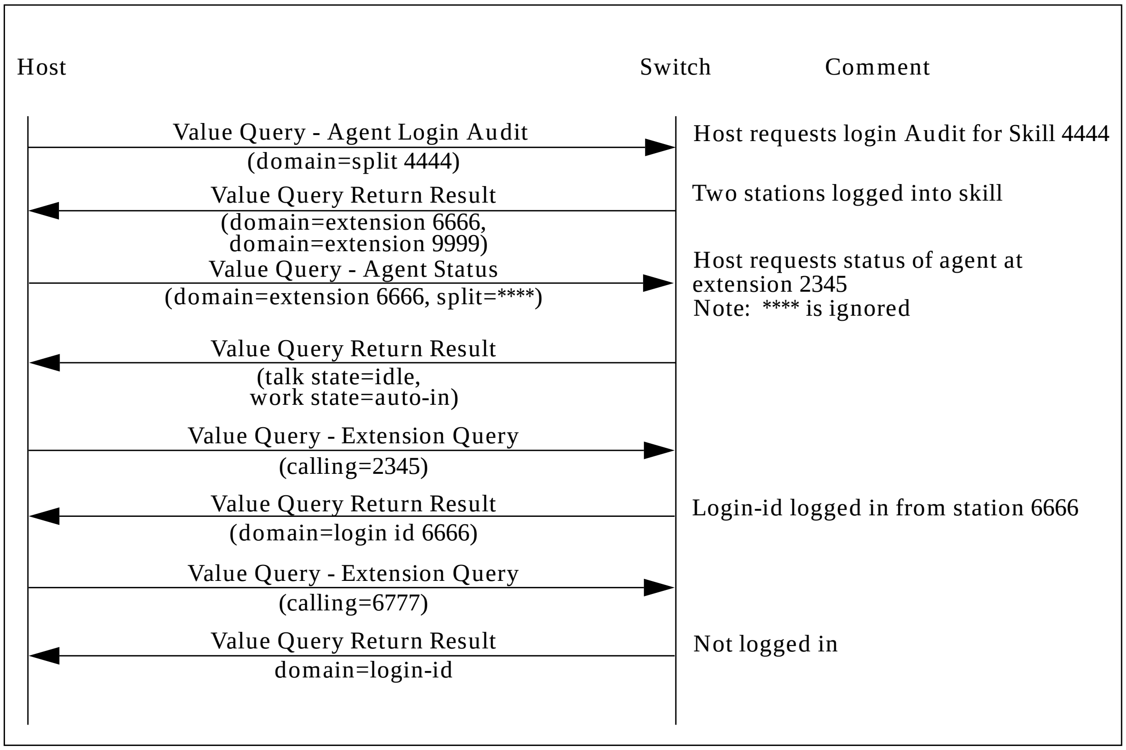


Figure A-9. Skill Hunt Groups and Logical Agents


Host
Switch
Comment
Value Query - Agent Login Audit
(domain=split 4444)
Host requests login Audit for Skill 4444
Value Query Return Result
(domain=extension 6666,
domain=extension 9999)
Two stations logged into skill
Value Query - Agent Status
(domain=extension 6666, split=****)
Host requests status of agent at
extension 2345
Note: **** is ignored
Value Query Return Result
(talk state=idle,
work state=auto-in)
Value Query - Extension Query
(calling=2345)
Value Query Return Result
(domain=login id 6666)
Login-id logged in from station 6666
Value Query - Extension Query
(calling=6777)
Value Query Return Result
domain=login-id
Not logged in

4. Converse Vector Command Interactions

External Call to a VDN that has a Converse Step that is Interrupted

This scenario presents the call flow for an incoming ISDN PRI call for VDN 7000 that has a Converse vector command that can be interrupted (see Figure A-10). The call comes into the VDN and gets queued to two splits, Split 6500 and Split 3400. The converse vector command then sends the call to the VRU (Split 1234) while maintaining the call's position in the other queues. When an agent in Split 6500 becomes available, the call leaves the VRU and is delivered to the agent. This ‘‘transfer’’ happens regardless of whether or not the caller has completed the VRU interaction.

Note that the Alerting Event Report sent when the call alerts the VRU port contains a cause value — CS3/23 (call remains in queue). This cause value informs the application that this is a converse split and that the call will not lose its place in any other splits that it has been queued to.

VDN 7000 has Event Notification active and each port on the VRU has Domain Control active.

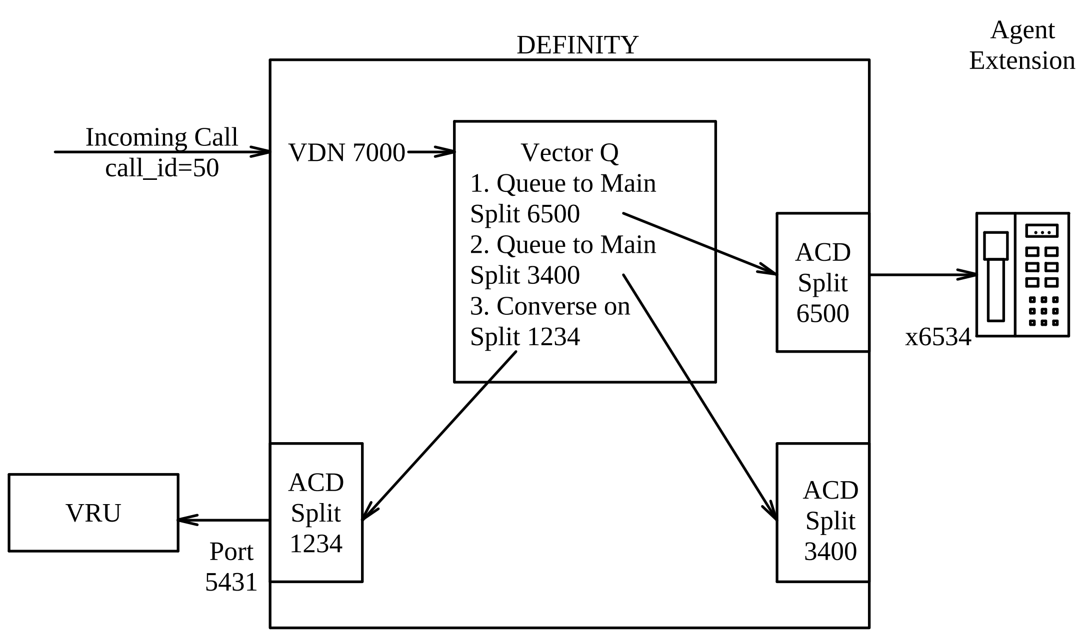
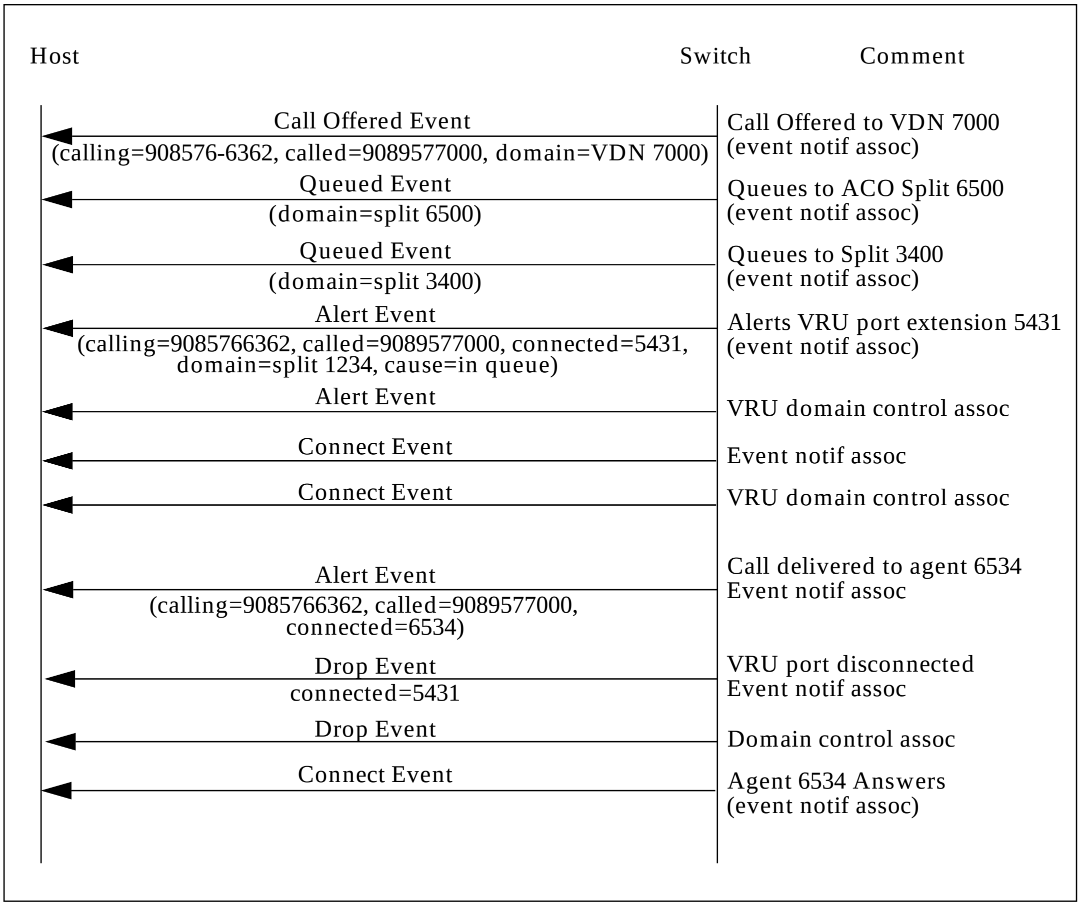


Figure A-10. Call Flow for a Converse Step that can be Interrupted


Host
Switch
Comment
Call Offered Event
(calling=908576-6362, called=9089577000, domain=VDN 7000)
Call Offered to VDN 7000
(event notif assoc)
Queued Event
(domain=split 6500)
Queues to ACO Split 6500
(event notif assoc)
Queued Event
(domain=split 3400)
Queues to Split 3400
(event notif assoc)
Alert Event
(calling=9085766362, called=9089577000, connected=5431,
domain=split 1234, cause=in queue)
Alerts VRU port extension 5431
(event notif assoc)
Alert Event
VRU domain control assoc
Connect Event
Event notif assoc
Connect Event
VRU domain control assoc
Alert Event
(calling=9085766362, called=9089577000,
connected=6534)
Call delivered to agent 6534
Event notif assoc
Drop Event
connected=5431
VRU port disconnected
Event notif assoc
Drop Event
Domain control assoc
Connect Event
Agent 6534 Answers
(event notif assoc)

External Call to a VDN that has a Converse Step that is not Interrupted

This scenario presents the call flow for an incoming ISDN PRI call for VDN 7001 that has a converse vector command that is not interrupted (see Figure A-11). The converse vector command passes both the ANI and the VDN number to the VRU. The VRU, after completing the session with the caller, sends the call back to vector processing. Along with sending the call back, the VRU also sends data back to the DEFINITY. This data is collected in a collected digits step. An adjunct route is then done that sends these collected digits to the ASAI Adjunct Processor. The ASAI Adjunct Processor then routes the call to ACD Split 3456.

Note that in this scenario, vector processing requires the caller to complete the interaction with the VRU before any additional processing is done to the call. Furthermore, the Alerting Event Report sent when the call alerts the VRU port contains a cause value — CS3/23 (remains in queue). This is to inform the ASAI Adjunct Processor that this is a converse split.

VDN 7001 has Event Notification active and each port on the VRU has Domain Control active.

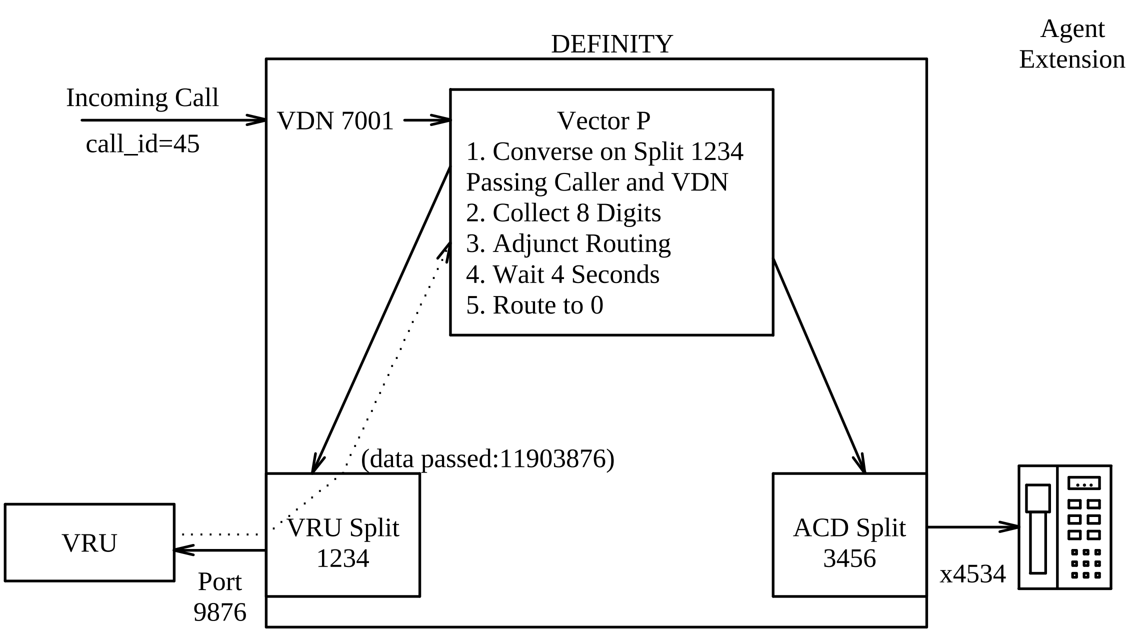
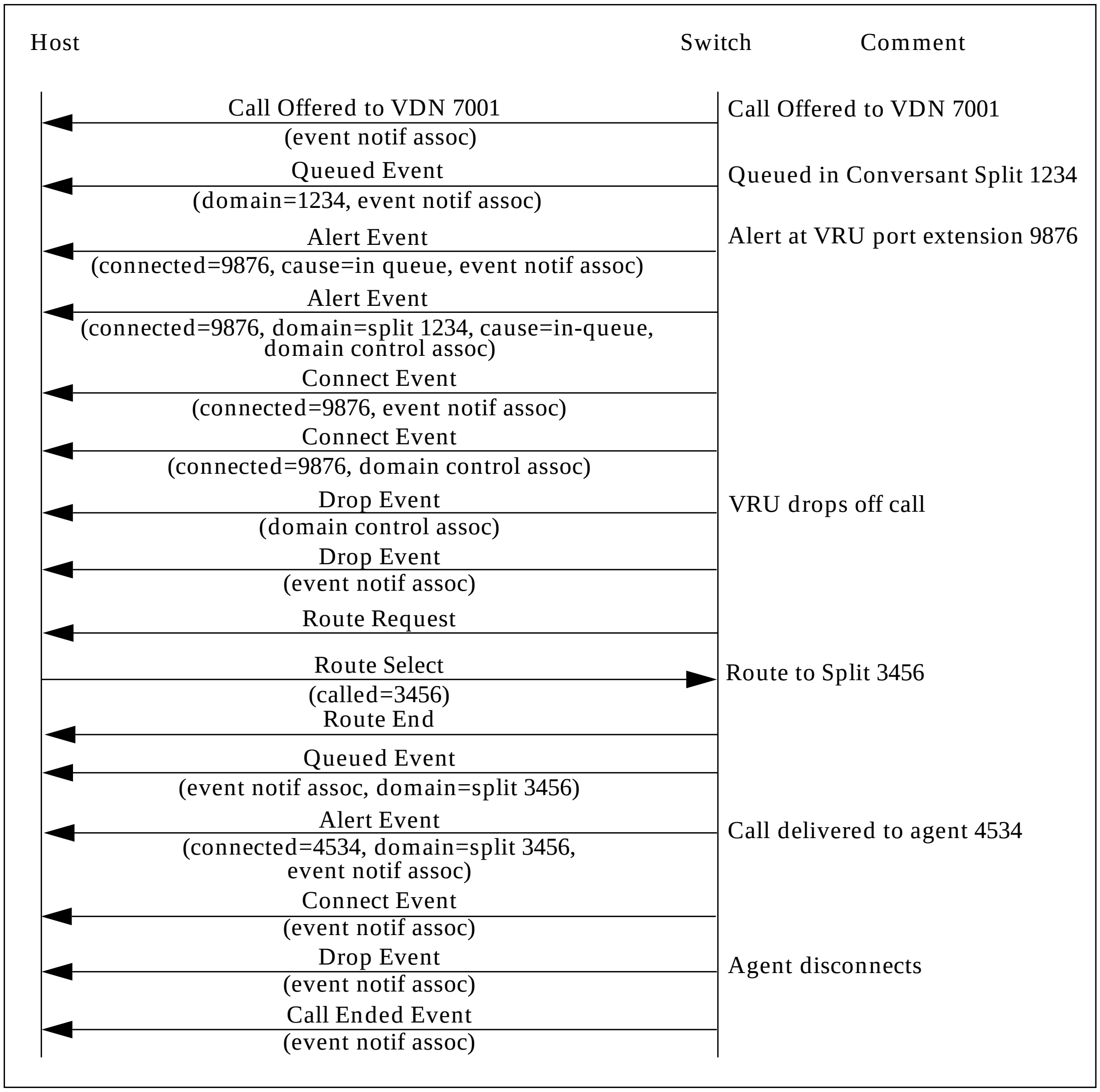


Figure A-11. Call Flow for a Converse Step that will not be Interrupted


Host
Switch
Comment
Call Offered to VDN 7001
(event notif assoc)
Call Offered to VDN 7001
Queued Event
(domain=1234, event notif assoc)
Queued in Conversant Split 1234
Alert Event
(connected=9876, cause=in queue, event notif assoc)
Alert at VRU port extension 9876
Alert Event
(connected=9876, domain=split 1234, cause=in-queue,
domain control assoc)
Connect Event
(connected=9876, event notif assoc)
Connect Event
(connected=9876, domain control assoc)
Drop Event
(domain control assoc)
VRU drops off call
Drop Event
(event notif assoc)
Route Request
Route Select
(called=3456)
Route to Split 3456
Route End
Queued Event
(event notif assoc, domain=split 3456)
Alert Event
(connected=4534, domain=split 3456,
event notif assoc)
Call delivered to agent 4534
Connect Event
(event notif assoc)
Drop Event
(event notif assoc)
Agent disconnects
Call Ended Event
(event notif assoc)

5. Redirection On No Answer (RONA) Interactions

Call to Agent with RONA

This scenario shows an incoming ISDN PRI call to VDN 7010 that is delivered to extension 6534 in split 6500 (see Figure A-12). The call is not answered by the agent at extension 6534 before the RONA timer expires. When the timer expires, the call is requeued to split 6500 and delivered to agent's station 6540.

In addition, extension 6534 is placed on AUX-work when the RONA timer expires so that no more ACD calls are delivered to the extension. If the call was sent to an Auto-Available Split (AAS) and the AAS agent or port did not answer, RONA would have taken the agent's extension out of service by automatically logging out the extension that did not answer. If the AAS split has Domain Control active, the switch sends a Logout Event Report for the extension logged out.

VDN 7010 has Event Notification active over CRV 96.

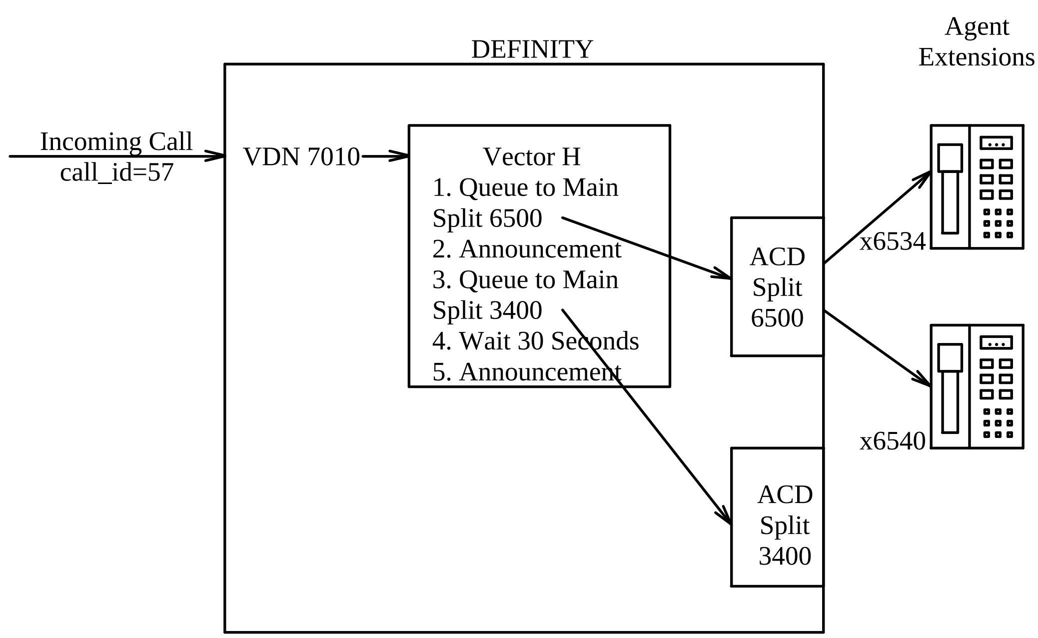
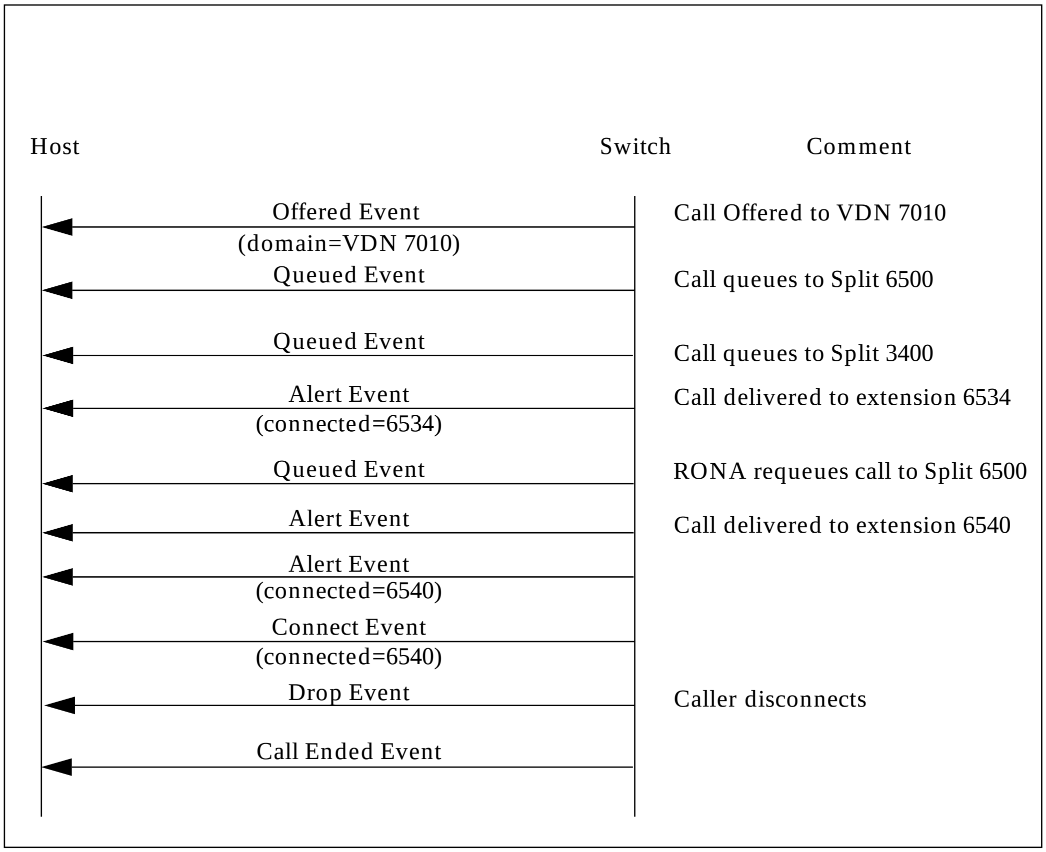


Figure A-12. Call Flow for a Call where RONA Timer Expires


Host
Switch
Comment
Offered Event
(domain=VDN 7010)
Call Offered to VDN 7010
Queued Event
Call queues to Split 6500
Queued Event
Call queues to Split 3400
Alert Event
(connected=6534)
Call delivered to extension 6534
Queued Event
RONA requeues call to Split 6500
Alert Event
Call delivered to extension 6540
Alert Event
(connected=6540)
Connect Event
(connected=6540)
Drop Event
Caller disconnects
Call Ended Event

Direct-Agent Call with RONA

This scenario presents the call flow for an incoming ISDN PRI call to VDN 8905 that gets routed, via direct-agent call, to extension 1234. The call is not answered by the agent at extension 1234 before the RONA timer expires (see Figure A-13). Because this is a direct-agent call, RONA will redirect the call to the agent's coverage path. Furthermore, the agent's extension will be placed in the AUX-work mode so that no more ACD calls will be delivered to the agent's extension.

If the incoming call was sent to an Auto-Available Split (AAS) and the agent (or port) selected did not answer before the RONA timer expired, the call would have been redirected back to the split (and queued at the highest priority) for distribution.

Note that an Agent Status Value Query on Extension 1234 was done by the adjunct processor prior to selecting that agent to receive the call. At that point in time, extension 1234 was in the Auto-In mode and in the idle talk state. A second agent status Value Query was done after the call was redirected away from extension 1234. This time extension 1234 is in the AUX-work mode and in the idle talk state.

Extension 1234 has Domain Control active over CRV 102. VDN 8905 has Event Notification active over CRV 96 and VDN 9876 is not monitored. Extension 1234 is logged into ACD split 1200.

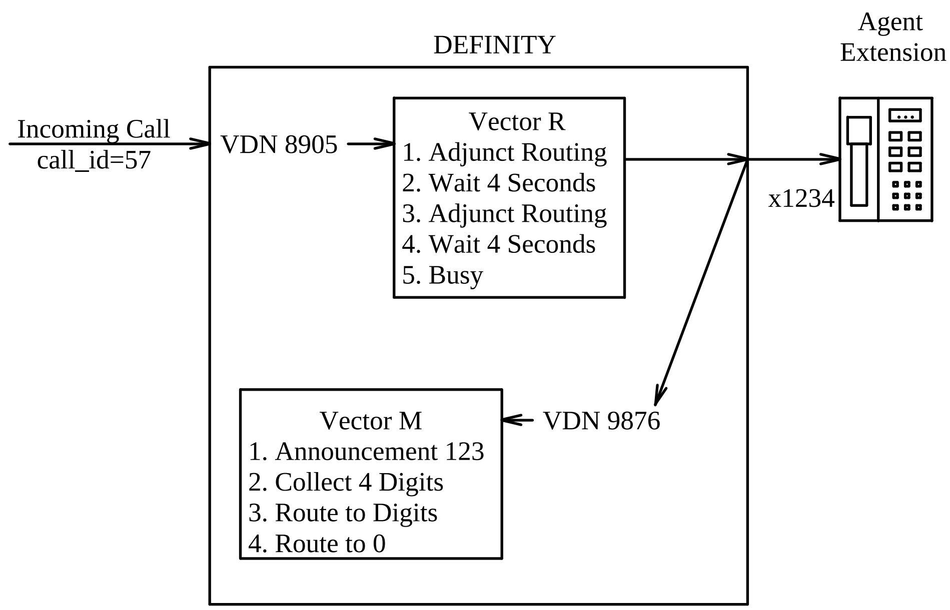
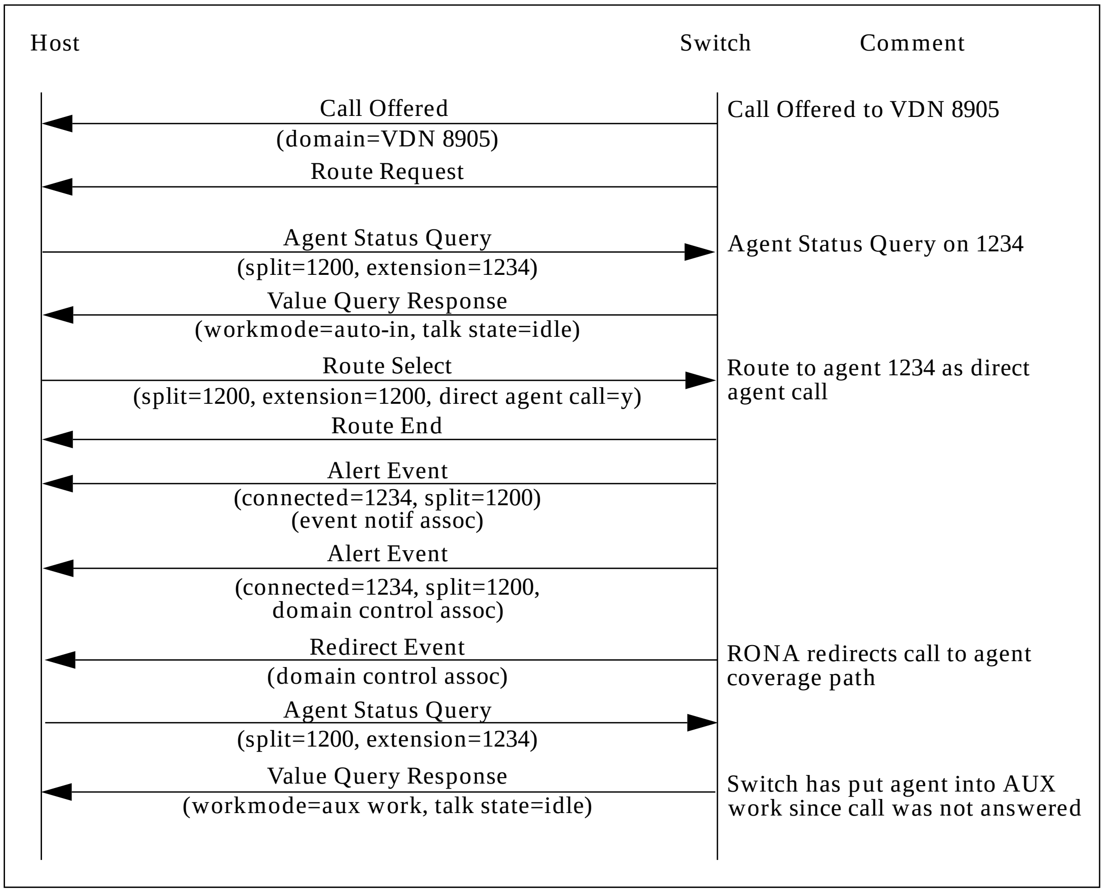


Figure A-13. Call Flow for a Direct Agent Call where RONA Timer Expires


Host
Switch
Comment
Call Offered
(domain=VDN 8905)
Call Offered to VDN 8905
Route Request
Agent Status Query
(split=1200, extension=1234)
Agent Status Query on 1234
Value Query Response
(workmode=auto-in, talk state=idle)
Route Select
(split=1200, extension=1200, direct agent call=y)
Route to agent 1234 as direct agent call
Route End
Alert Event
(connected=1234, split=1200)
(event notif assoc)
Alert Event
(connected=1234, split=1200,
domain control assoc)
Redirect Event
(domain control assoc)
RONA redirects call to agent coverage path
Agent Status Query
(split=1200, extension=1234)
Value Query Response
(workmode=aux work, talk state=idle)
Switch has put agent into AUX work since call was not answered

6. VDN in Coverage Path Interactions

Incoming Call Routed to a Station that has a VDN in the Coverage Path

This scenario shows the call flow for an incoming non-ISDN call that gets routed to extension 1234 via the **adjunct routing** command. Extension 1234 does not answer the call and the call covers to extension 9876. Extension 9876 does not answer the call and the third coverage point is VDN 3634 (see Figure A-14).

VDN 8905 has Event Notification active over CRV 96. Extensions 1234 and 9876 have Domain Control active over CRV 80 and 95, respectively.

The ASAI messages generated by the **adjunct routing** vector command are also shown.

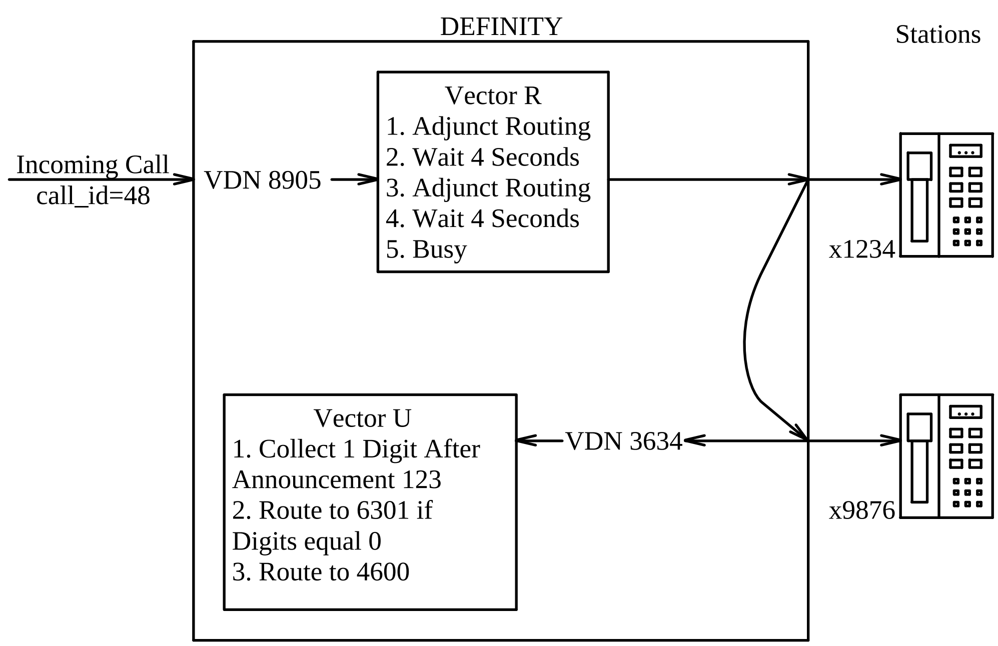
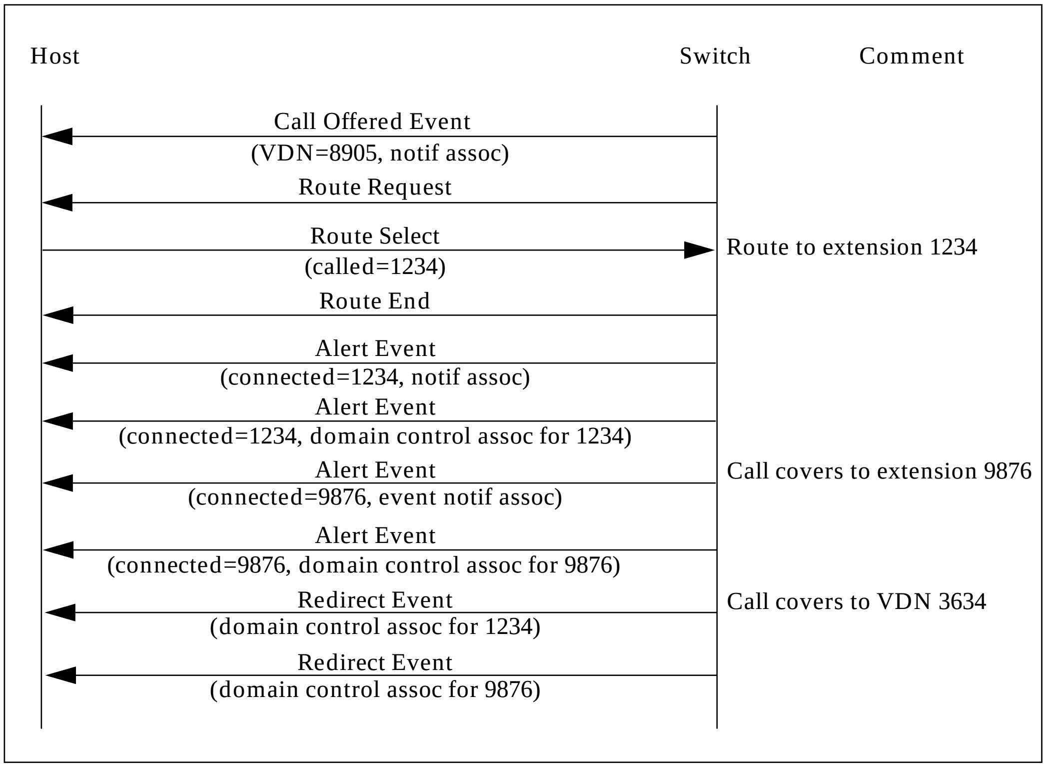


Figure A-14. Call Flow for an Agent who has a VDN in the Coverage Path


Host
Switch
Comment
Call Offered Event
(VDN=8905, notif assoc)
Route Request
Route Select
(called=1234)
Route to extension 1234
Route End
Alert Event
(connected=1234, notif assoc)
Alert Event
(connected=1234, domain control assoc for 1234)
Alert Event
(connected=9876, event notif assoc)
Call covers to extension 9876
Alert Event
(connected=9876, domain control assoc for 9876)
Redirect Event
(domain control assoc for 1234)
Call covers to VDN 3634
Redirect Event
(domain control assoc for 9876)

External Call to a VDN with a Forced First Announcement that gets Routed to a Second VDN

This section presents the call flow for an incoming ISDN PRI call for VDN 5678 that hears a forced first announcement (See Figure A-15). After the announcement, the call gets routed via the **adjunct routing** vector command to VDN 5700. The call eventually gets answered by Agent 4566 in Split 3460.

Note that no event reports are generated for the announcement. In general, ACD split forced first or second announcements and vector-controlled announcements do not generate announcements. However, event reports are generated for non-split announcements.

VDN 5678 has Event Notification active over CVS 98.

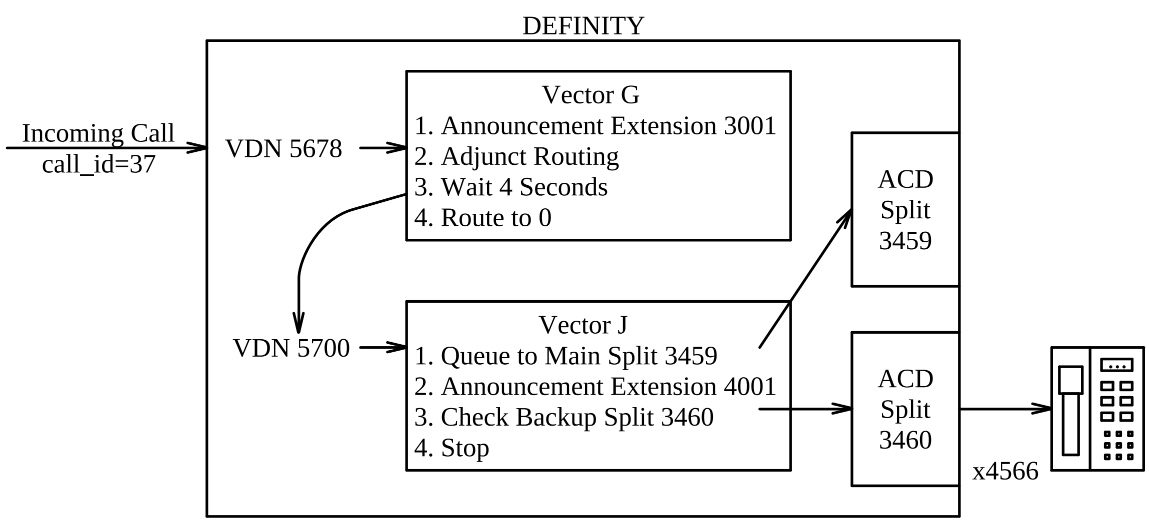
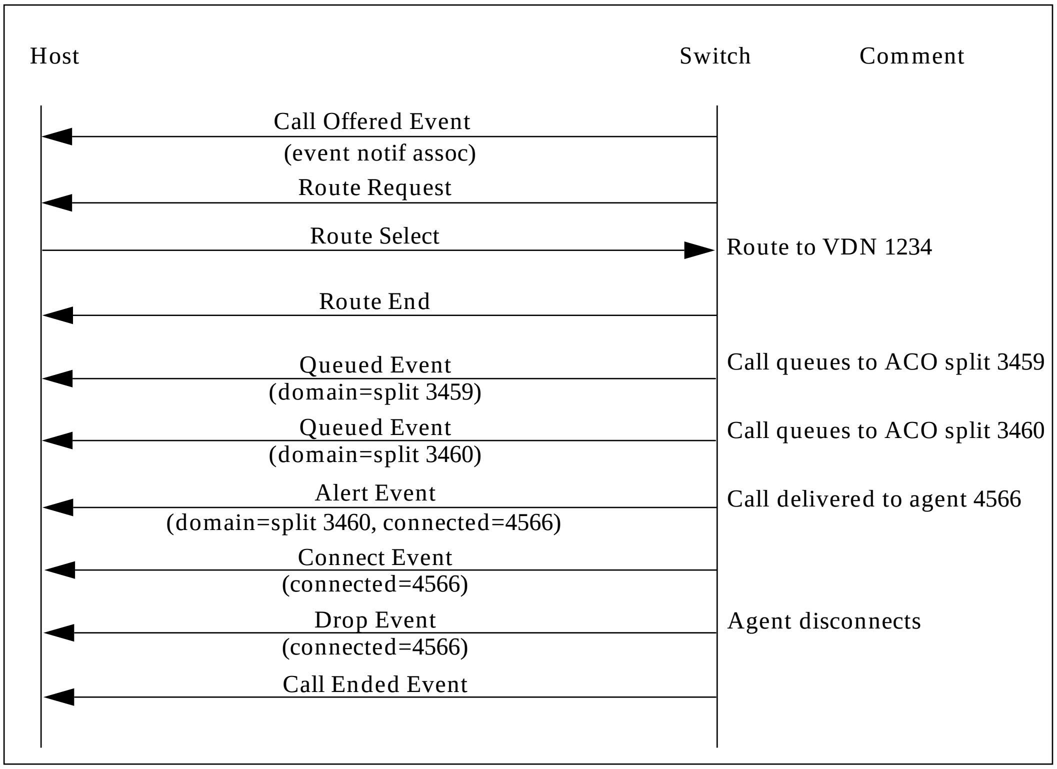


Figure A-15. Call Flow for Call to a VDN with Announcement and Routed to Another VDN


Host
Switch
Comment
Call Offered Event
(event notif assoc)
Route Request
Route Select
Route to VDN 1234
Route End
Queued Event
(domain=split 3459)
Call queues to ACO split 3459
Queued Event
(domain=split 3460)
Call queues to ACO split 3460
Alert Event
(domain=split 3460, connected=4566)
Call delivered to agent 4566
Connect Event
(connected=4566)
Drop Event
(connected=4566)
Agent disconnects
Call Ended Event

Outgoing Call over Non-ISDN Trunk

This section presents the call flow for an outgoing call over a non-ISDN trunk. Station 1234 initiates this preview dialing call (see Figure A-16).

Note that a Trunk Seized Event Report is generated when the switch places the call over a non-ISDN trunk. Furthermore, no Alerting or Connected Event Reports follow a Trunk Seized Event Report. The only event report that may be generated for the destination is a Dropped Event Report.

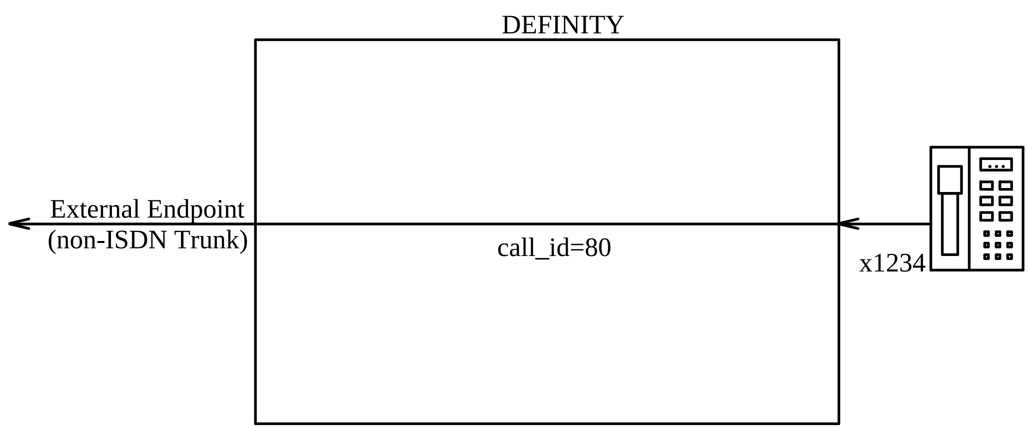
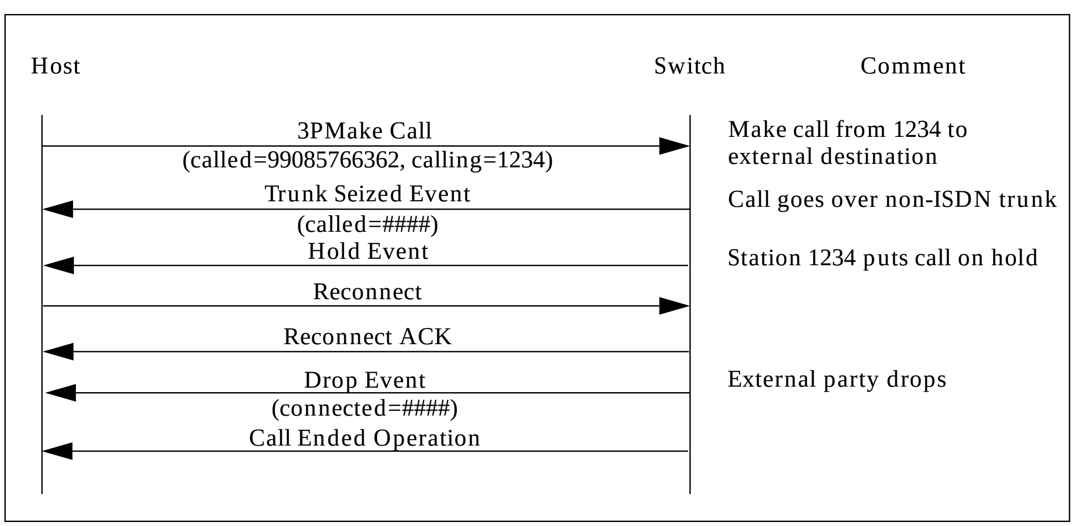


Figure A-16. Outgoing Call over Non-ISDN Trunk



Outgoing Call over ISDN Trunk that Results in an ISDN Progress Message

This section presents the call flow for an outgoing call over an ISDN trunk that traverses a non-ISDN network(s) before it reaches its destination. Station 4567 initiates an outgoing call to an external destination. Station 4567 has Domain Control active and uses the Auto Dial capability to initiate the call (see Figure A-17).

When a call leaves the ISDN network, an ISDN PROGress message is sent from the PRI network to the switch and subsequently to the ASAI Adjunct Processor. The switch sends the contents of the PROGress message in a Cut-Through Event Report. Multiple PROGress messages may be sent for a call; each one is mapped into a Cut-Through Event Report.

For a call that has resulted in a Cut-Through Event Report being generated, the Alerting Event Report is optional. A Connected and/or Drop Event Report is always sent as long as the call utilizes the ISDN facilities.

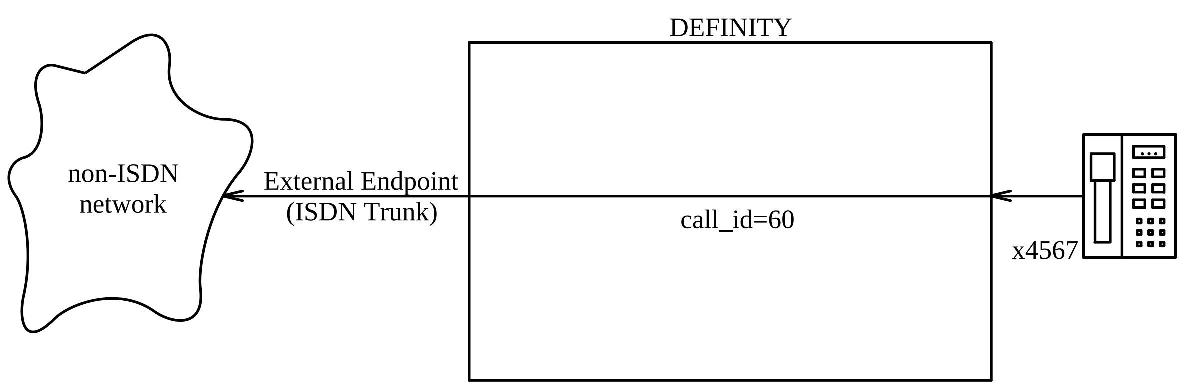
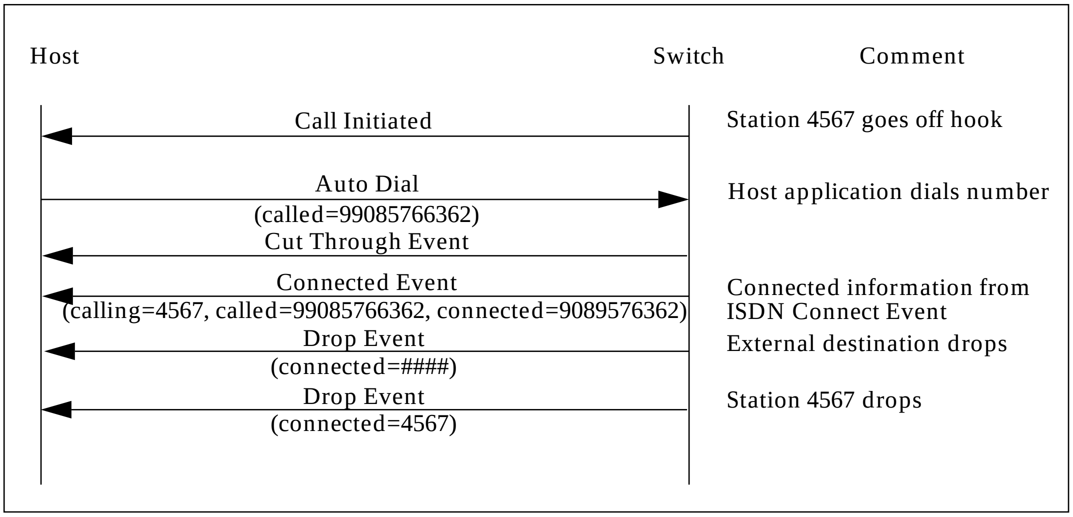


Figure A-17. Call Flow for Outgoing ISDN Call that Traverses a non-ISDN Network


Host
Switch
Comment
Call Initiated
Station 4567 goes off hook
Auto Dial
(called=99085766362)
Host application dials number
Cut Through Event
Connected Event
(calling=4567, called=99085766362, connected=9089576362)
Connected information from
ISDN Connect Event
Drop Event
(connected=####)
External destination drops
Drop Event
(connected=4567)
Station 4567 drops

7. User Scenarios — User to User Information (UUI)

Figure A-18 shows a typical distributed call center configuration. An ASAI host is connected to each of the switches and calls are delivered to either switch. The applications running on the ASAI hosts are not connected to each other.

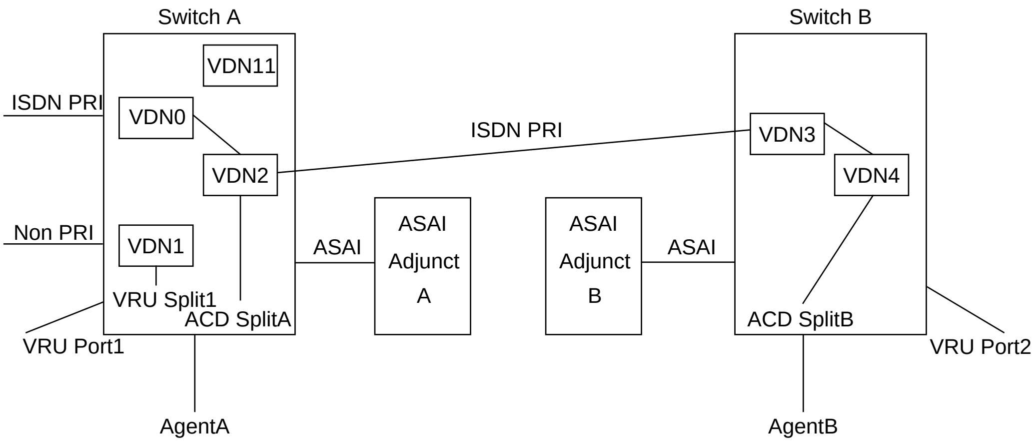


Figure A-18. Distributed ACD Configuration

The following call sequences show typical UUI scenarios:

Call Sequence 1 shows an incoming non-ISDN PRI call to switch A, delivered to VDN1. VDN1 delivers the call to VRU Port1 in VRU Split1. After the caller terminates the VRU session, the ASAI host transfers the call to VDN2 for further processing. The ASAI host includes UUI in the Third Party Make Call used to start the transfer. While the call is in VDN2, the call interflows to switch B. The call is accepted at switch B and an announcement is played while the call waits in queue for ACD SplitB.

Call Sequence 1:

Host A	Host B	Comment
Call Offered (called=VDN1, domain=VDN1, trunk group=AAA)		Incoming call to Switch A
Alert Event (connected=VRU Port 1, domain=split VRU Split 1)		Call delivered to VRU
Connect Event (connected=VRU Port 1, domain=split VRU Split 1)		Call connected to VRU
Hold Event (connected=VRU Port 1)		VRU places call on hold
3PMake Call Request (UUI=info1, called=VDN2)		VRU requests New Call to do transfer to VDN2
Call Transferred Event		VRU transfers call to VDN2
Queued Event (called=VDN2, domain=ACD Split A)		
		Call intraflows to Switch B (VDN3 is the LAI VDN; VDN4 is the VDN for accepted calls)
	Call Offered (called=VDN3, calling, domain=VDN4, LAI info)	
	Queued Event (called=VDN3, domain=ACD split 3)	
Connect Event (called=VDN3)		Call connects to announcement: answer supervision reported on trunk
	Alert Event (called=VDN3, connected=agentB, domain=splitB, UUI=info1)	
	Connected Event (called=VDN3, connected=agentB)	
		Agent B disconnects
	Drop Event (connected=agentB)	
Call Ended Event	Call Ended Event	

Call Sequence 2 shows an incoming ISDN PRI call to switch A, delivered to VDN1. VDN1 contains a collect digits vector step followed by an adjunct routing vector step. The host routes the call, including UUI information, to VDN2 that tries to interflow the call to switch B. Switch B does not accept the call and the call connects to Agent A in switch A.

Call Sequence 2:

Host A	Host B	Comment
Call Offered (called=VDN1, domain=VDN1)		Incoming call to Switch A
		Call Prompting Collects Digits
Route Request (called=VDN0, collected=001)		
Route Select (called=VDN2, UUI=info2)		
Route End		
Queued Event (called=VDN2, domain=ACD SplitA)		
	Call Offered (domain=VDN3, UUI=info2, LAI info)	Call intraflows from Switch A to B (VDN3 is the LAI VDN)
Drop Event (connected=####)	Call Ended Event	Switch B denies call
Alert Event (called=VDN1, connected=agentA, domain=ACD splitA, UUI=info2)		Call delivered to agent
Connected Event		
Drop Event (connected=agentA)		Agent drops
Call Ended Event		

Call Sequence 3 shows an incoming ISDN PRI call to switch A, delivered to VDN1. The incoming ISDN call contains UUI data. While in VDN1, the call is routed to VDN3 in switch B including UUI information and a return call destination (VDN11). Switch B connects the call to VRU Port2. After the VRU terminates the caller session, the ASAI host drops the call including UUI information back to switch A. When the trunk to switch B drops, the call is directed to VDN11. The host on switch A drops the call including UUI information in the request.

Call Sequence 3:

Host A	Host B	Comment
Call Offered (called=VDN1, UUI=info0)		
Route Request (called=VDN1, domain=VDN0, UUI=info0)		
Route Select (called=VDN3, UUI=info3, returncall dest=VDN11)		
Route End		
	Call Offered (called=VDN3, domain=VDN3, UUI=info3)	
Alert Event (called=VDN3, connected=####)	Alert Event (called=VDN3, connected=VRU port 2, UUI=info3)	Call delivered to VRU
Connected Event (called=VDN3, connected=####)	Connected Event (called=VDN3, connected = VRU port 2)	
	3P Drop Request (UUI=info4)	VRU disconnects, provides UUI
Drop Event (connected=####, UUI=info4)	3P Drop ACK	
	Call Ended Event	
		Call delivered to Return VDN
Call Offered (called=VDN1, domain=VDN11, UUI=info4)		
3P Drop Request (UUI=info5)		Host drops call
3P Drop ACK		
Call Ended		

8. User Scenarios — Connected IE for Non-ISDN Trunks

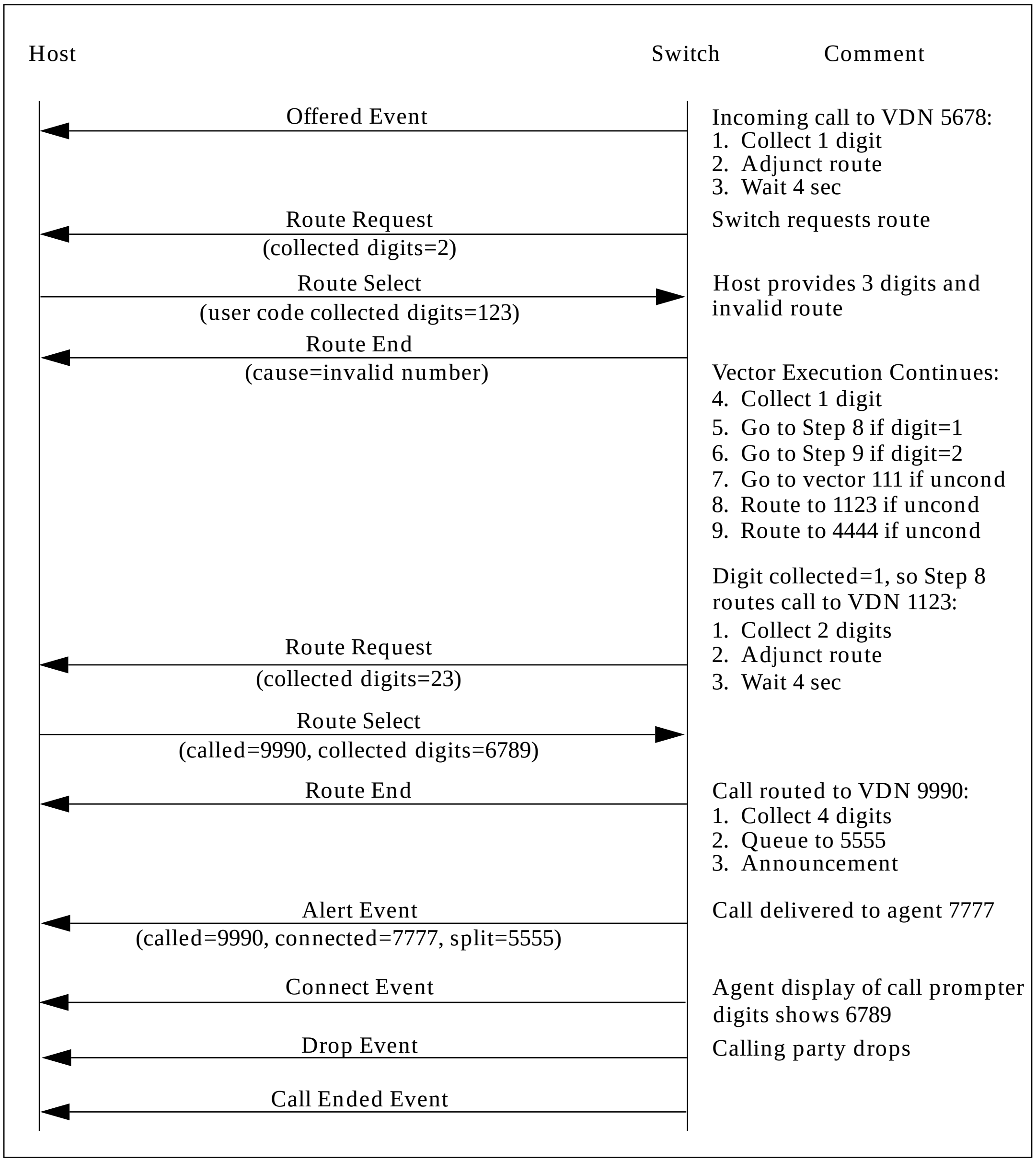
Table A-1 shows the Event Reports provided for a monitored call that is routed over an outgoing non-ISDN trunk. The incoming call also uses a non-ISDN trunk and is directed to a VDN/vector that routes the call to an external number.

Table A-1. Incoming Call Routed to External Destination Example

Operation	Event Report
Incoming Call	Call Offered call_id=45 trunk group=102 called number=65678 domain=VDN 65678
Call Routed to External Destination	
Non-ISDN Trunk Seized	Trunk Seized call_id=45 party_id=2 called number=#####
Call Connected Answer Supervision Received from the Network (or timed by switch)	Connect call_id=45 party_id=2 trunk group=102 called number=##### connected number=##### cause=normal
Called Party Drops	Drop call_id=45 party_id=2 cause=normal connected number=#####
Call Terminates	Call Ended call_id=45 cause=normal

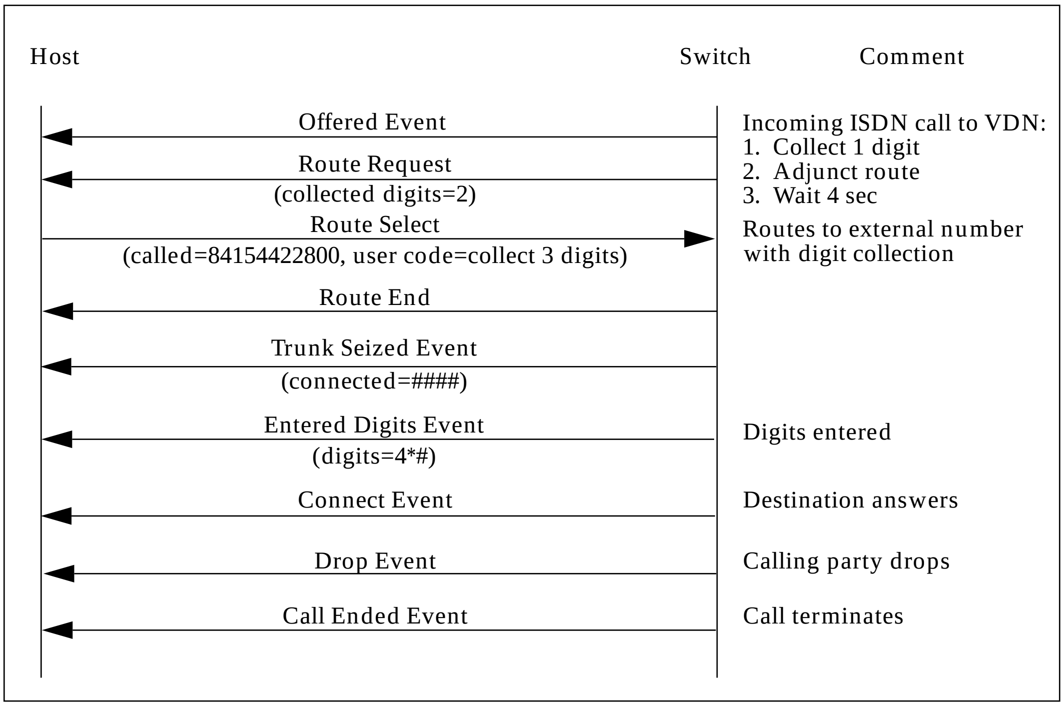
9. User Scenarios — ASAI-Provided Dial-Ahead Digits

This is a simple scenario in which the host provides dial-ahead digits via a Route Select. After the dial-ahead digits are stored by the switch, the digits are collected using call prompting vector commands. The scenario also shows the ASAI Event Reports sent to a monitoring host.



10. User Scenarios — ASAI-Requested Digit Collection

This is a sample scenario for an incoming ISDN call that is routed via Adjunct Routing to an external destination. The user has subscribed to receive 4-digit DNIS numbers. As part of the route, the host requests collecting three digits from the caller.



11. User Scenarios — VDN Return Destination

A customer may use the VDN Return Destination feature (G3V2 and later) to provide a more flexible remote access feature together with host-based call security. The remote user/caller does not have to call back into the switch when multiple destinations need to be reached or enter his/her identification every time a new destination is desired. For example, a customer can program the following vector that is accessed by dialing a VDN that has a Return Destination administered.

1. Collect 8 digits after announcement 1001 (‘‘Please enter your identification number and password followed by # sign.’’)
2. Adjunct Routing link extension XXX1
3. Wait 6 seconds hearing silence.
4. Collect 16 digits after announcement 1002 (‘‘Please enter the telephone number of your destination followed by # sign.’’)
5. Adjunct Routing link extension XXX1
6. Wait 6 seconds hearing silence.
7. Disconnect after announcement 1003 (‘‘We are sorry, but we are experiencing technical difficulties at this time, please try again later.’’)

In this scenario, a remote caller calls into the switch by dialing the VDN administered with the Return Destination. The vector executed prompts the caller to enter an identification number and a password that will be passed, via the **adjunct routing** vector command, to the host for validation. The host can keep track of invalid attempts or decide to de-activate or activate certain identification numbers based on customer set criteria.

After the host-based security is passed, the switch collects digits for the destination the caller wants to reach (vector step 4 above). The host receives the number entered by the caller (vector step 5 above) and validates the entered number to check if the caller is allowed to reach the specified destination. If so, the host routes the call to the desired (dialed) destination.

If the host security is not passed, the host routes the call to an appropriate alternate destination (for example, announcement with security violation message) and log the invalid call attempt. If the host is not available, the call is disconnected after an announcement (vector step 7 above).

After the called destination disconnects from the call, the caller can remain on the line to be connected to the Return Destination. A sample Return Destination vector is as follows:

1. Collect 16 digits after announcement 1002 (‘‘Please enter the telephone number of your next call followed by # sign.’’)
2. Adjunct Routing link extension XXX1

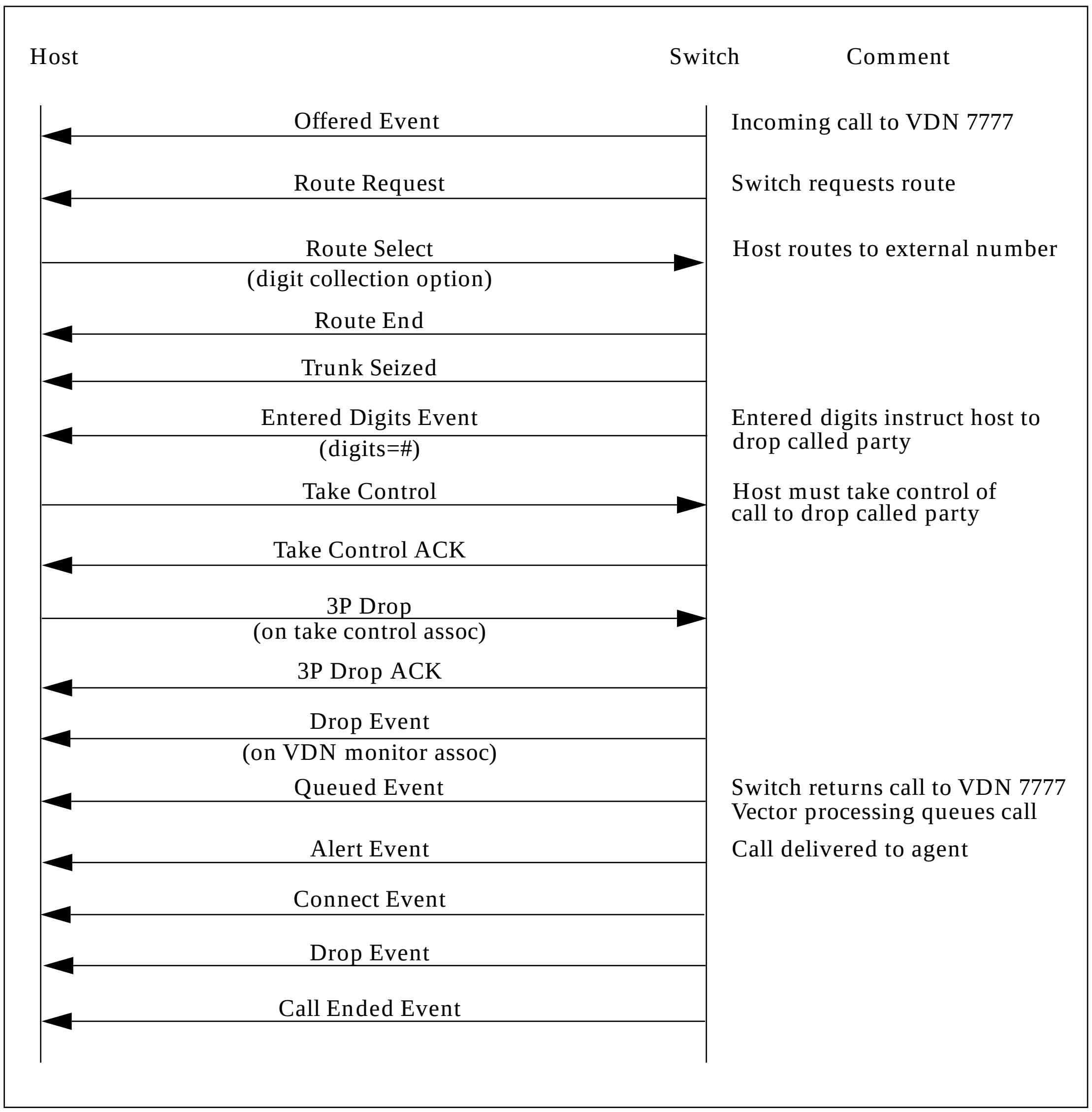
3. Wait 6 seconds hearing silence.
4. Disconnect after announcement 1003 (‘‘We are sorry, but we are experiencing technical difficulties at this time, please try again later.’’)

Once connected to the Return Destination, the caller can enter a second destination/phone number to connect to. The host performs the same validation on the destination number as in the first destination and routes the call as appropriate (destination entered by caller or alternate destination). Note that the host can also provide reports on all the destinations and times reached by each remote user.

In the Return Destination vector, it is recommended that the first vector command give the caller the opportunity to disconnect from the call rather than immediately routing the call to some destination. If the call was immediately routed and then the caller decided to hang-up, the destination that the call was routed to would ring, alerting the called party, but then no one would be on the line at the other end (this could be confusing to customers, and could be misinterpreted as a problem with the feature). Vector commands such as **wait**, **collect after announcement**, and **announcement** can provide the caller with the opportunity to disconnect before the call is routed. As an example, an **announcement** command with the recording ‘‘Please hang-up to end your call, or remain on the line if you wish to place another call’’ instructs the caller to disconnect before the call is routed.

12. ASAI Messaging Scenarios — VDN Return Destination

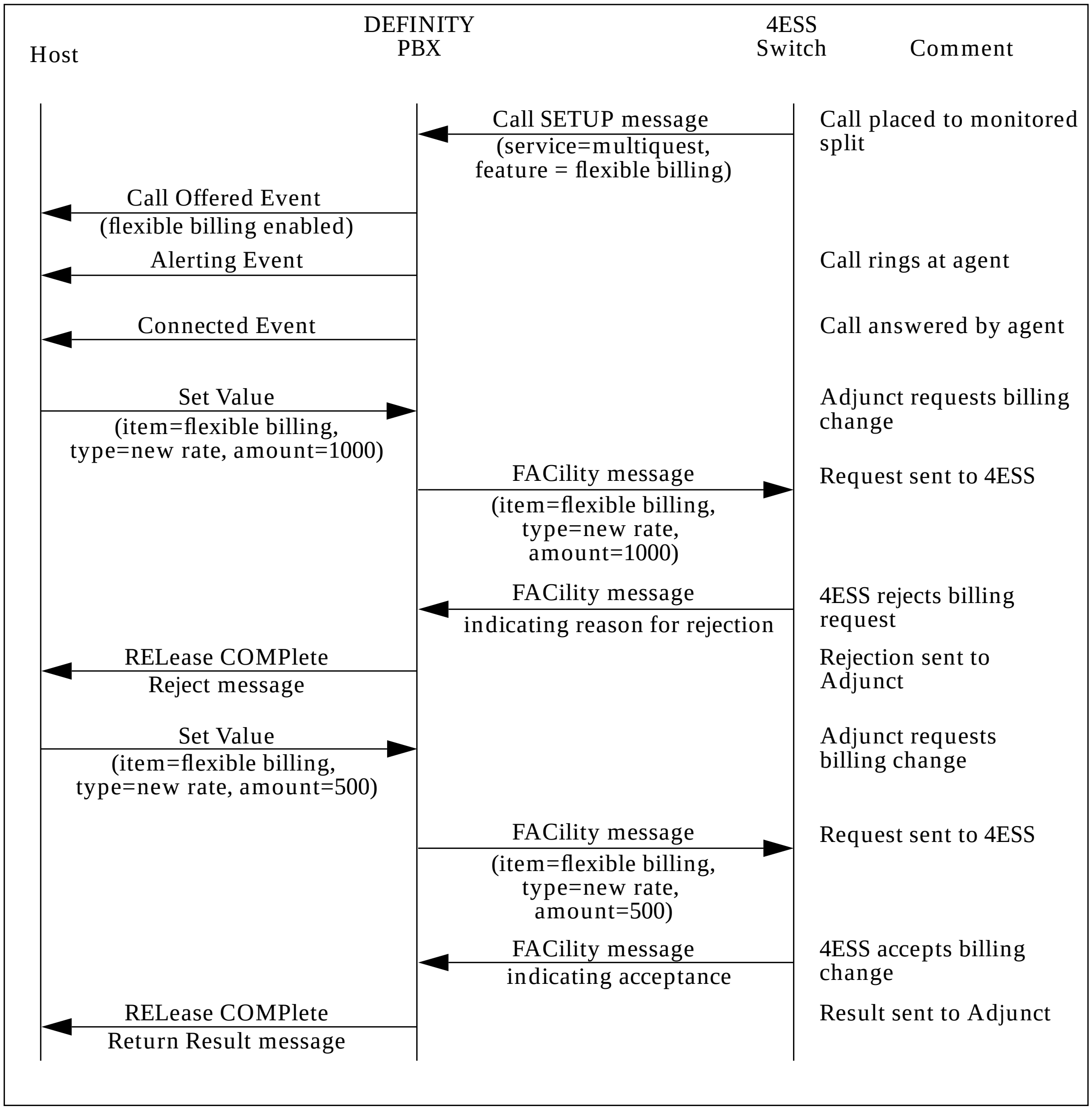
This is a scenario where a call to a vector is routed using Adjunct Routing to an external destination. The host then drops the external destination and the call is delivered to the Return Destination for further vector processing. The scenario assumes that the call is being monitored by the ASAI host and that the Return Destination is VDN 77777.



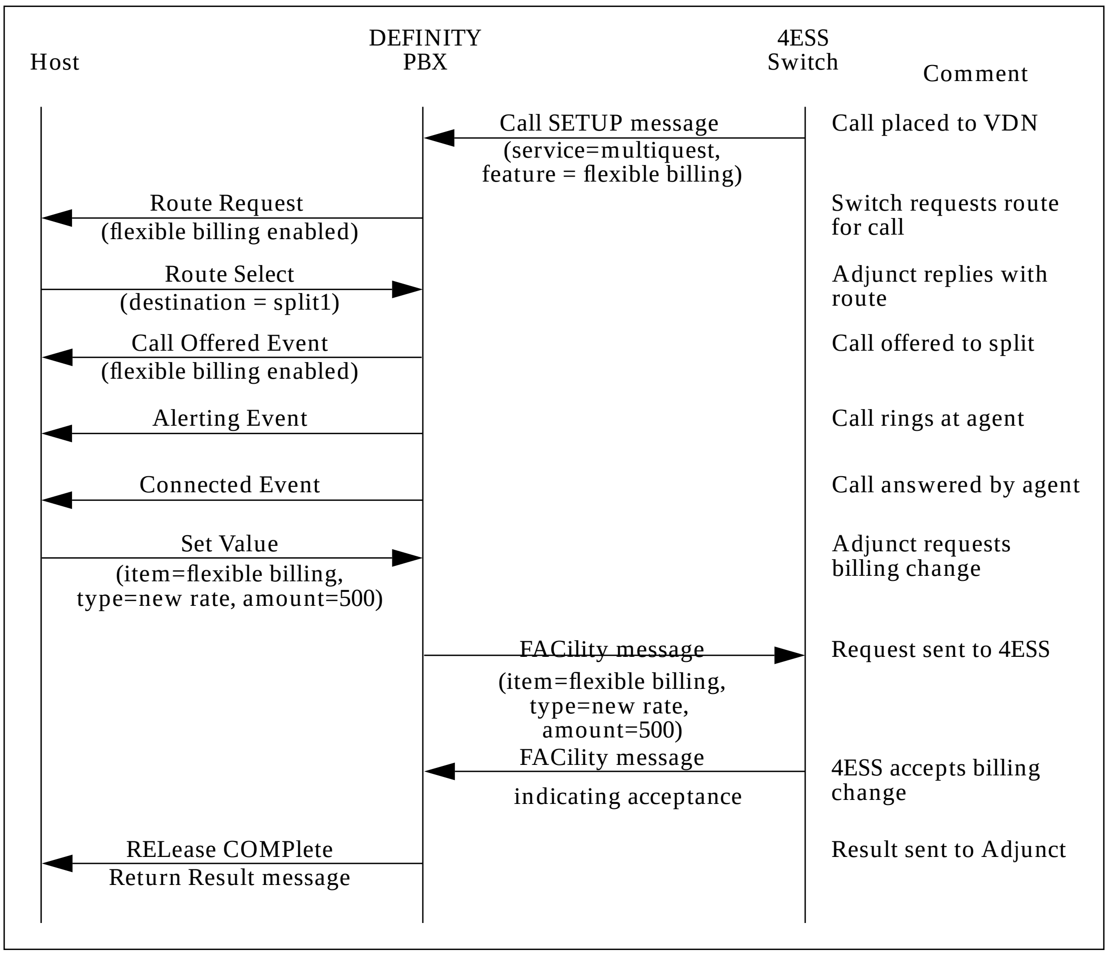
13. User Scenarios — Flexible Billing

The following call sequences show typical Flexible Billing scenarios.

Call Sequence 1 shows an incoming call on an ISDN-PRI trunk delivered to Split A. The ISDN trunk is configured for MultiQuest service. The incoming call indicates in its SETUP message that Flexible Billing is supported. The call rings at agent 5001 and is answered. The agent requests a billing change on the call, setting the new rate to $10/minute. This billing change is sent over the ISDN-PRI trunk, and the 4ESS rejects the change. The switch sends the response to the ASAI adjunct. The agent requests a billing change on the call, setting the new rate to $5/minute. This billing change is sent over the ISDN-PRI trunk, and the 4ESS accepts the change. The switch sends the response to the ASAI adjunct.

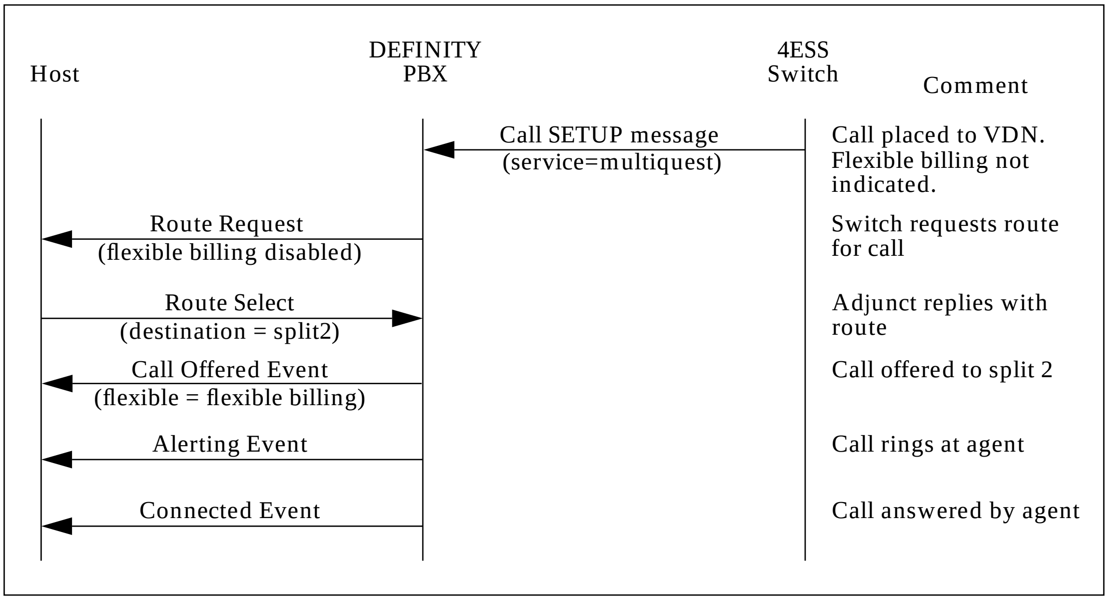


Call Sequence 2 shows an incoming call on an ISDN-PRI trunk delivered to VDN A. The ISDN trunk is configured for MultiQuest service. The incoming call indicates in its SETUP message that Flexible Billing is supported. The call is adjunct-routed. The adjunct sees that Flexible Billing is enabled, and routes the call to split 1. The call rings at agent 5001 and is answered. The switch sends the response to the ASAI adjunct. The agent requests a billing change on the call, setting the new rate to $5/minute. This billing change is sent over the ISDN-PRI trunk, and the 4ESS accepts the change. The switch sends the response to the ASAI adjunct.



Call Sequence 3 shows an incoming call on an ISDN-PRI trunk delivered to VDN A. (This call sequence uses the same VDN as call sequence 2.) The ISDN trunk is configured for MultiQuest service. The incoming call indicates in its SETUP message that Flexible Billing is NOT supported. The call is adjunct-routed. The adjunct sees that Flexible Billing is NOT enabled and routes the call to split 2. The call rings at agent 5001 and is answered.

In scenario 2, the agent knew that Flexible Billing was available on the call since the call routed to split 1. In scenario 3, the agent knew to bill by a means other than Flexible Billing since the call routed to split 2.



ASAI and Generic 3 Switch Requirements

B

This appendix provides capacity requirements and constraints for ASAI on the Generic 3 switch (G3i and G3r).

Capacity Requirements and Constraints

Table B-1. System-Wide ASAI Limits

Capacity	G3s	G3i	G3r
ASAI Links	4	8	8
Notification Associations	50	170	2,000
Adjunct-Control Associations	75	300	3,000
Domain-Control Station Associations	250	2,000	6,000
Domain-Control Split Associations	99	99	255
Simultaneous Call-Class Originators	20	40	400
Simultaneous Billing Change Requests	25	100	1,000
Simultaneous Intern. Meas. Data Queries	25	25	50

- System Limits
  - — The **maximum number of ASAI links** is four for G3s, eight for G3i, and eight for G3r.
  - — The **maximum number of active-notification associations** is 50 for G3s, 170 for G3i, and 2,000 for G3r.

— The **maximum number of adjunct-control associations** is 75 for G3s, 300 for G3i, and 3,000 for G3r.[1] This number consists of the total active Third Party Make Call (or Third Party Take Control Calls) plus any in-progress Third Party Call Control requests (made over a domain control association). It also includes outstanding adjunct routing requests as well as outstanding feature and value query requests.

— The switch sets aside internal resources for the **maximum number of domain-control station associations** (250 for G3s, 2,000 for G3i, and 6,000 for G3r). When multiple domain-control associations (maximum of two) are controlling the same station, each association uses resources from the pool. Thus, when there are two controllers per station, then the number of stations that may be domain-controlled is 125, 1,000, or 3,000, respectively.

— The switch sets aside internal resources for the **maximum number of domain-control split associations** (99 for G3s and G3i, and 255 for G3r), which is the maximum number of hunt groups.

— The **maximum number of call-classifier originators** is 20 for G3s, 40 for G3i, and 400 for G3r. This number determines the maximum number of outbound switch-classified calls using a call classifier resource that may be in progress simultaneously. Such calls require the call classifier until the call is classified (destination answers or call is dropped), and then the switch releases the classifier. This number does not limit the number of connected outbound switch-classified calls active at a given time.

— The **maximum number of simultaneous billing change requests** is 25 for G3s, 100 for G3i, and 1000 for G3r.

— The **maximum number of simultaneous internally measured data queries** is 25 for G3s, 25 for G3i, and 50 for G3r.

- **Per station domain:**

Table B-2. ASAI Limits per Station

Capacity	G3s	G3i	G3r
Domain Controllers per Station Domain	2	2	2

1. The limitation may be particularly relevant for OCM applications. Value queries would be made as part of the call pacing algorithm and the Third Party Make Call capability would actually place the calls.

- — The maximum number of active domain-control associations per station is two.
- — The maximum number of associations that can control a station on a call is three, two domain-control associations and one call-control association.

- **Per ACD split domain:**

Table B-3. ASAI Limits per ACD Split Domain

Capacity	G3s	G3i	G3r
Active Notifications per Split Domain	3	3	3
Domain Controllers per Split Domain	1	1	1

- — The maximum number of active notifications per split domain is three.
- — The maximum number of domain controllers is one.

- **Per VDN domain:**

Table B-4. ASAI Limits per VDN Domain

Capacity	G3s	G3i	G3r
Active Notifications per VDN Domain	3	3	3

- — The maximum number of active notifications per VDN domain is three.

- **Per call:**

Table B-5. ASAI Limits per Call

Capacity	G3s	G3i	G3r
Adjunct-Control Associations per Call	1	1	1
Active Notif. Associations per Call	3	3	3
Domain Control Associations	12	12	12

- — The maximum number of adjunct-control associations per call is one.
- — The maximum number of active-notification associations per call is three.

- — The maximum number of domain-control associations per call is 12 (since the maximum number of stations on a call is six and each may have two domain control associations).
- — The maximum number of associations that can control a call is 13, two domain-control associations per station on a call and one adjunct-control association (2x6+1).
- — Typically, the maximum number of associations that can receive event reports for a single call is 16 (13 Controlling and three active-notification). The exception to this rule is the transfer event. Up to 18 associations may be sent a transfer event.

■ **Per ASAI interface (per link):**

Table B-6. ASAI Limits per Link

Capacity	G3s	G3i	G3r
CRVs switch-to-adjunct	127	127	127

- — The maximum number of simultaneous ASAI associations that the switch supports in the switch-to-adjunct direction on any link is 127 (for example, there cannot be more than 127 pending routing requests on an ASAI link).
- — The maximum number of simultaneous associations supported in the other direction (for example, adjunct-to-switch) depends on the length of the ASAI CRV value. The maximum length of the ASAI CRV value field is two bytes. An administration option lets a customer set the length of the CRV value to 1 or 2 bytes. The maximum number of associations that an adjunct can initiate and have active on an ASAI interface at any one time depends on how this parameter is administered:
  - ► If the CRV values are administered to a single byte, then the adjunct may initiate and have active 127 simultaneous associations on any given ASAI interface (subject to availability of resources set by the other system parameters).
  - ► If the CRV values are set at two bytes, the maximum number of associations is limited by the availability of other switch resources, not the number of CRVs (32,767). That is, the maximum number of associations that the switch can support over all ASAI interfaces (combined) is less than the number of available CRVs when a 2-byte CRV length is administered. The demands on other switch resources prevent an adjunct from ever reaching the theoretical maximum of 32,767.

Glossary

GL

ACD Automatic Call Distribution

ACD agent/extension A station extension that is a member of an Automatic Call Distribution (ACD) split/skill/hunt group.

ACD call A call directed to an ACD split/skill/hunt group either directly or through vector processing.

ACD split/skill/hunt group A specific type of hunt group that distributes similar type calls to the available agent/station extension that has been idle for the longest time.

Active call For the Send DTMF Signals feature, a call that has received answer supervision, either network or timed (that is, resulting from elapse of a software timer), and has assigned listen and talk time slots. Therefore, for Send DTMF Signals purposes, an answered call on hold is an active call.

Active-Notification Association A ‘‘link’’ initiated by the adjunct allowing it to receive Event Reports for a specific switch entity; for example, an outgoing call. This association is initiated by the adjunct via the *Event Notification Request* capability.

Active-Notification Call
A call for which Event Reports are being sent over an active-notification association (communication channel) to the adjunct. Sometimes referred to as a ‘‘monitored call.’’

Active Notification Domain
A VDN or ACD Split extension for which Event Notification has been requested.

Adjunct
See **Application.**

Adjunct-Control Association
A relationship initiated by an application via the *Third Party Make Call*, *Third Party Take Control*, or *Domain (Station) Control* capabilities to set calls up and control calls already in progress.

Adjunct-Controlled Call
Includes all calls that can be controlled using an adjunct-control association. These calls must have been originated via the *Third Party Make Call* or *Domain (Station) Control* capabilities, or must have been taken control of via the *Third Party Take Control* or *Domain (Station) Control* capabilities.

Adjunct-Controlled Split
An ACD split administered to be under adjunct control. Agents logged into such splits must do all telephony and ACD login and/or logout and change work mode functions through the adjunct (except for auto-available adjunct controlled splits, whose agents may not be logged in and/or logged out or have their work modes changed).

Adjunct-Monitored Call
Includes all adjunct-controlled calls, active-notification calls, and calls that provide event reporting over domain-control associations.

Adjunct Processor
Also called Application Processor. A customer-provided processor used in conjunction with ASAI for call monitoring and control. *See also* **ASAI Host/Adjunct Processor**.

Adjunct Routing
A vector command/step that allows the switch to request a route/destination, from an ASAI adjunct, for the call executing the vector command/step. When an **adjunct routing** vector command is encountered, the switch disconnects any tone detector/call prompter connected to the call and discards any dial-ahead digits that had been collected.

Collected digits are retained with the call and sent to the adjunct in the Route Request message.

Announcement
An administered extension that provides a recorded message. DEFINITY supports both internal (announcement circuit pack - TN750) and external (TIE trunk connected to recording device; for example, A15 Unit) announcement sources.

Answer Supervision
A signal sent by a terminating communication system to an originating communication system, or intermediate charging point, such as a central office (CO) switch, indicating that an incoming call has been answered. Upon receiving this signal, the originating system or other charging point begins tracking charges for the call, if charges apply.

In terms of network services, answer supervision is a feature offered by the network provider on certain types of trunks.

AP
Adjunct Processor (or Application Processor).

Application
An adjunct entity that requests and receives ASAI services or capabilities. One or more applications can reside on a single adjunct. However, the switch cannot distinguish among several applications residing on the same adjunct and treats the adjunct and all resident applications as a single application. The terms ‘‘application’’ and ‘‘adjunct’’ are used interchangeably. *See also* **ASAI Application.**

Application Processor
See **Adjunct Processor, ASAI Host/Adjunct Processor**.

ASAI

Adjunct Switch Application Interface (ASAI).

1. The AT&T recommendation for Computer Telephony Integration (CTI) based on the CCITT Q.932 protocol.

2. An option on the DEFINITY switch that enables the ASAI messaging interface. Also called CallVisor ASAI.

Adjunct Services Application Interface (ASAI).

1. A messaging interface between the switch and an Adjunct Processor that allows the AP to perform call monitoring and control functions.

ASAI Application

An application running on an ASAI adjunct by making calls to a library written to meet the ASAI specifications. *See also* **Application**.

ASAI Host/Adjunct Processor

A computer processor that communicates with the switch via an ASAI link. *See also* **Application Processor**.

ASAI link

An ISDN BRI or Ethernet interface configured to support ASAI.

Association

A communication channel between the adjunct and switch for messaging purposes. An active association is an existing call on the switch or an extension on the call.

Or, a single instance of an ASAI capability group (for example, Third Party Call Control, Notification, Third Party Domain Control, Routing) between an ASAI adjunct and the switch.

Also, a virtual relationship established between the switch and the AP used to relate messages and events to a particular call or to an ASAI capability. An association is represented by a unique Call Reference Value (CRV)/link combination.

Auto-available Split

A specific type of ACD split/hunt group that automatically logs in its members and places them in auto-in mode as soon as the switch is initialized or when a member is added to the auto-available split. Normally used when dedicated equipment answers the call directed to the split. Also, allows the movement of agents to other splits without the agent being logged out.

Automatic-answer (auto-answer)
A feature that allows an agent to receive ACD calls while off-hook.

Available Agent
An agent available to receive a call through an ACD split (ACD call); that is, an agent who/which is logged in and not on a call, and in the auto-in or manual-in work mode.

Basic Call Management System
Switch features that provide a subset of the functionality of a CMS.

BCMS
Basic Call Management System

Billing Number (BN)
The 10-digit number (North America) that is billed when the calling party makes a toll call. The BN is not always identical to the Calling Party Number (CPN); for example, a company may have a BN of 555-7000, yet an individual at the company may have a CPN of 555-7335. Taken together, the CPN/BN information allows the called party to identify the calling party and determine call volumes from particular geographic areas. *See also* **Calling Party Number (CPN)**.

BN
Billing Number

BRI
The ISDN Basic Rate Interface specification.

Call Control Capabilities
Capabilities (*Third Party Selective Hold, Third Party Reconnect, Third Party Merge*) that can be used in either of the Third Party Call Control ASE (cluster) subsets: Call Control and Domain Control.

Calling Party Number (CPN)
The 10-digit number (North America) of the station that is calling. The CPN is not always identical to the billing number (BN); for example, a company may have a BN of 555-7000, yet an individual at that company may have a CPN of, for example, 555-7335. Taken together, CPN/BN information allows the called party to identify the calling party and to determine call volumes from particular geographic areas. Formerly called station identification (SID). *See also* **Billing Number** (BN).

Call Management System (CMS)	An application that collects, stores, analyzes, displays and reports ACD information provided by the switch. CMS enables customers to monitor and manage telemarketing centers by generating reports on the status of agents, splits, trunks, trunk groups, vectors, and vector directory numbers, and enables customers to partially administer the ACD features on the switch.
Call Prompting	A feature that uses vector commands to collect and test digits, and/or display digits to the agent or pass them to an adjunct. Based on the digits dialed, the call is routed to a desired destination or receives other treatment. External callers must have touch-tone dialing for entering digits. Call Prompting can be used with other call vectoring capabilities.
Call Reference Value (CRV)	An identifier present in ISDN messages that associates a related sequence of messages. In ASAI, the CRVs distinguish between associations.
Call Vectoring	A method that manages inbound calls, using routing tables to uniquely define treatments for each call type. The call type is based on the dialed number or trunk group termination to a vector via vectoring directory numbers. The vectors are customer-programmable using commands that resemble a high-level programming language to specify what treatments the call should be given. Also called ''vectoring.'' *See also* **Vectors** and **Vector step**.
Capability	Either a request for or an indication of an operation. For example, a *Third Party Make Call* is a request for setting up a call and an *Event Report* is an indication that an event has occurred.
Capability Groups	Sets of capabilities that denote association types. For example, *Call Control* is a type of association that allows certain functions (the ones in the capability group) to be performed over this type of association. Each capability group may contain capabilities from several capability groups. Groups are provisioned

through switch administration, and can be requested by an application.

Referred to in other documentation as administration groups or Application Service Elements (ASEs).

Cause Value
A value returned in responses to requests or in event reports when a denial occurs or an unexpected condition is encountered. ASAI cause values fall into two ''coding standards'': Coding Standard 0 includes cause values that are part of AT&T and CCITT ISDN specifications, and Coding standard 3 includes any other ASAI cause values. ASAI documents use a notation for cause value where the coding standard for the cause is given first, followed by a slash and the cause value. For example, CS0/100 is coding standard zero, cause value one hundred.

Class of Restriction
A feature that allows definition of up to 96 classes of call-origination and call-termination restrictions for telephones, telephone groups, data modules, and trunk groups.

Collected digits
Touch-tone digits entered by a caller and collected by a **collect digits** vector step. *See also* **dial-ahead digits**.

CMS
Call Management System

Controlled Call
A call for which an ASAI adjunct has Third Party Call Control. Controlled calls include calls controlled via Third Party (Single Call) Call Control and Third Party Domain (Station) Control associations.

Controlled Station
A station being monitored and controlled via a domain-control association.

CONVERSANT VIS
An AT&T Voice Response Unit product often used with the DEFINITY switch to perform call screening, redirection, and data collection functions.

Converse session
The period of time during which a call is under the control of a converse vector step and the calling party is interacting with a Voice Response Unit (VRU) or ACD agent.

Converse split	A split/skill or hunt group accessed by a converse vector step. (The term ''non-converse split'' is used to signify any split/skill or hunt group accessed by a queue to main or check backup vector step.)
COR	Class of Restriction
Coverage call	A call that is redirected from the called extension to another extension or group of extensions when certain criteria are met. *See also* **Coverage criteria**.
Coverage criteria	The conditions under which a call to a principal is redirected to coverage — for example, the Send All Calls button is on, the line is busy, or all the appearances are in use. Also called ''criteria'' and ''redirection criteria.'' *See also* **Coverage call**.
CPN	Calling Party Number
CRV	Call Reference Value
DCS	Distributed Communications System
Denying a Request	Sending a negative acknowledgement (NAK) or Facility Information Element (FIE) with a *return error* component. (A cause value is also provided). This should not be confused with the Denial Event Report, which applies to calls.
Dial-ahead digits	Touch-tone digits entered by a caller but not yet collected by a **collect digits** vector step. Dial-ahead digits are stored in the call prompting buffer until collected by a **collect digits** vector step. *See also* **Collected digits**.
Distributed Communications System	A private network of PBXs in which some features are transparent across PBXs. Or, a network configuration linking two or more switches in such a way that selected features appear to operate as if the network were one system/switch.

Domain	An entity that can be controlled or monitored.
Domain-Control Association	The unique ‘‘CRV/link number’’ combination initiated by a *Third Party Domain Control Request* capability.
Domain-Controlled Split	A split for which *Third Party Domain Control* request has been accepted. A domain-controlled split provides an event report for logout.
Domain-Controlled Station	A station for which a *Third Party Domain Control* request has been accepted. Provides event reports for calls that are alerting, connected, or held at the station.
Domain-Controlled Station on a Call	A station active on a call that provides event reports over one or two domain-control associations.
DTMF	Dual Tone Multi-Frequency
EAS	Expert Agent Selection
Expert Agent Selection	An optional feature that allows call center agents to have assigned skills and to receive calls based on their skill. EAS adds flexibility to ACD. Each agent is assigned to a station at login time. This makes it possible to use the same physical station for a variety of skills and agents. Or, an ACD feature in which calls can be directed to specialized pools of agents who possess the correct skills to handle the call.
FAC	Feature Access Code
Facilities	Trunks connecting the DEFINITY switch to public or private networks.
Facility IE	A Q.931 Codeset 6 Information Element containing information to be passed between communications entities. This IE can be included in several Q.931 messages and, as a parameter, contains additional IEs within itself.

FACility Message	A Q.931 message sent during an ASAI association to convey information from one endpoint to another as part of the message exchange for that ASAI association. Also used to pass change rate requests for the Flexible Billing feature to the 4ESS switch and to receive success or failure responses from the 4ESS switch.
Feature Access Code	A 1-, 2-, 3-, or 4-digit dial code used to activate or cancel a feature or to access an outgoing trunk. Star (*) or pound (#) can be used as the first digit of an access code.
Flexible Billing	A feature that allows ASAI to change the rate at which an incoming 900-type call is billed.
IE	Information Element
IMD	Internally Measured Data
Information Element (IE)	A defined and identifiable structure within an ISDN message that contains particular information relevant to the call and that message. An ISDN message consists of an appropriate set of IEs, some mandatory and others optional.
Internally Measured Data	A feature that allows an ASAI adjunct to query for the data items available to station displays via VuStats.
Logical Agent	A logical extension assigned to an agent when the EAS feature is enabled on the switch. A logical agent logs into a skill, and by so doing, maps the logical extension to a physical extension.
Manual Answer	An operation in which an Automatic Call Distribution (ACD) agent is on-hook and available to receive an ACD call, the call comes in via ringing on the station set, and the agent goes off-hook on the ringing appearance to answer the call.
Manual-in Mode	An ACD work mode that indicates an agent is available to receive an ACD call. When the agent is in manual-in mode, the agent is automatically put into after-call work state on termination of the current ACD call.

Monitored call	A call that provides ASAI Event Reports over Event Notification associations.
Monitored domain	The VDN, split, or agent specified in an ASAI Event Notification Request. ASAI messaging to support this feature only applies to monitored domains.
NSF IE	Network-Specific Facility IE. Part of the ISDN SETUP message that contains the identification of the service used to process that particular call. This adds the ability to act upon a second optional NSF IE.
Outbound Dialing	The mechanism used by an ASAI adjunct to ask the switch to launch a call on behalf of an ACD split, detect when the call is answered, and connect the far end to an available agent in the split.
Party/Extension Active on Call	A party is on the call if it is actually connected to the call (in active talk or in held state). An originator of a call is always a party on the call. Alerting parties, busy parties, and tones are not parties on the call.
PCOL	Personal Central Office Line
PRI	Primary Rate Interface
Primary Extension	The main extension associated with the physical station set
Primary Rate Interface	A standard Integrated Services Digital Network (ISDN) frame format that specifies the protocol used between two or more communications systems. North American PRI runs at 1.544 Mbps and provides 23 64-Kbps B-channels (voice or data) and one 64-Kbps D-channel (signaling). The D-channel is the 24th channel of the interface and contains multiplexed signaling information for the other 23 channels.
Principal	A station that has its primary extension bridged on one or more other stations.
Queue	An ordered sequence of items, such as outgoing trunk calls, incoming Automatic Call Distribution (ACD) calls, or available agent positions waiting to be processed.

Queuing	The process of holding calls in order of their arrival to await connection to an attendant, answering group, or idle trunk.
Receiving Endpoint	For the Send DTMF Digits feature, an endpoint that can listen to DTMF signals on behalf of a sender. A receiving endpoint's listen path is connected to the sender's talk path. Also called ‘‘receiver.’’
Redirection on No Answer	A feature that redirects an unanswered, ringing, ACD split call back to the split after making the agent unavailable and notifying the Call Center manager.
REGister Message	The ISDN message type used to initiate the ASAI Set Value and Value Query capabilities.
RELease COMplete Message	The ISDN message type used to complete the ASAI Set Value and Value Query capabilities.
Reorder Tone	A fast-busy tone repeated 120 times a minute, indicating that at least one of the facilities required for a call, such as a trunk or a digit transmitter, was not available when the call was placed.
Ringback Tone	The audible signal heard at a calling telephone to indicate that the called party is being rung. In some contexts, ringback tone does not mean that the called party is receiving an audible signal.
RONA	Redirection on No Answer
ROSE	Remote Operations Service Element. A CCITT and ISO standard that defines a notation and services that support interactions between the various entities that make up a distributed application.
Send DTMF Signals	A feature that allows the DEFINITY switch to generate DTMF tones on a talk path when requested to do so by the adjunct application.
Sending Endpoint	For the Send DTMF Digits feature, an endpoint on whose behalf the DTMF signals are sent. Also called ‘‘sender.’’

Skill Hunt Group	A hunt group for EAS (Expert Agent Selection) that has the same attributes as a vector-controlled split.
Simulated Bridged Appearance	The same as a **temporary bridged appearance**. Allows the station user (usually the principal) to bridge onto a call answered by another party on its behalf.
Split	A group of agents organized to receive calls in an efficient and cost-effective manner.
Split night service	If a hunt group does not operate for certain hours, split night service provides the capability of forwarding new calls to a night destination but leaving the calls that are currently in queue until they have been handled. Night service for hunt groups allows different hunt and ACD groups to go into night service at different times. Night console service can also be administered on a trunk-group basis that has precedence over night console service for hunt groups.
Split supervisor	The manager of an Automatic Call Distribution (ACD) split who handles supervisor-assist calls.
Staffed	An Automatic Call Distribution (ACD) answering-position state indicating that the agent is present and logged in.
Supervisor assist	A feature that allows ACD agents to request assistance from their supervisors. The agent can confer with the supervisor, or transfer of conference the call to the supervisor. Also called ‘‘agent assist.’’
Timed collection	Digit collection with a 10-second interdigit time out. Every time a digit is received the interdigit timeout is reset and the tone detector/call prompter is disconnected if the interdigit time out expires. The **collect digits** vector command always executes timed collection of digits.
Timeout	The expiration of a preassigned time interval, during which a specified condition persisted. Timeout is normally associated with an automatic action — for example, in Loudspeaker Paging, after a timeout, the

	paging amplifiers and speakers are automatically released.
TEG	Terminating Extension Group
TAC	Trunk Access Code
To Control	The action an application takes when it invokes Third Party Call Control capabilities using either an adjunct-control or a domain-control association. *See* **Controlled call**.
To Monitor	The action an application takes when it receives Event Reports on either an active-notification, adjunct-control, or domain-control association. *See* **Monitored call**.
Transfer	A feature that allows a multifunction (digital or hybrid) telephone user to transfer a call by pressing the Transfer button, which places the current call on hold, calling or selecting the appearance of a third party, and completing the transfer by pressing the Transfer button a second time. Transfer may also be done via FAC or ASAI.
Unmonitored domain	A VDN, split, agent, or switch port not specified as a monitored domain. *See* **Monitored domain**.
Unstaffed	An Automatic Call Distribution (ACD) answering-position state indicating that the agent is not present.
User to User Information	An ISDN Q.931 Information Element used to carry user information transparently across an ISDN network. It is used by ASAI adjuncts to associate calls with adjunct/caller information during call routings and transfers.
UUI	User to User Information
VDN	Vector Directory Number
Vector command	A command used in call vectoring to specify the treatment a call will receive. Commands include main or backup Automatic Call Distribution (ACD) split queuing with priority levels and inflow-threshold checking; delays with specified feedback such as ringback, music, silence, or announcements; collecting

	digits; routing to internal or external destinations; and unconditional and conditional branching. Conditional branching is based on call-handling conditions of the ACD splits, collected digits, or on time of day and day of the week. *See also* **Vectors** and **Vector step**.
Vector-Controlled Split	A hunt group or ACD split administered with the ‘‘vector’’ field enabled. Access to such a split is only possible by dialing a VDN extension. Vector-Controlled Splits cannot be Active Notification Domains.
Vector Directory Number (VDN)	An extension that provides access to the Vectoring feature on the switch. Vectoring allows a customer to specify the treatment of incoming calls based on the dialed number. Or, an extension number that terminates at a vector. Calls to the VDN are processed by the vector to which the VDN points. When used with vectoring, the dialed number received by the communications system to provide Dialed Number Identification Service (DNIS) is the VDN assigned for that service.
Vectors	Easily programmed routing tables for processing incoming calls that provide various responses to the caller before the call is answered or receives other treatment. *See also* **Call vectoring**, **Vector Directory Number**, **Vector command**, and **Vector step**.
Vector step	One of a series of steps processed sequentially within a vector unless a step with a **Goto** command or **Stop** command is encountered or vector processing terminates. A step consists of an action to be taken and the information needed to complete the action. *See also* **Call vectoring**, **Vector command**, and **Vectors**.
Voice Response Unit	An adjunct product used to perform call screening, redirection, and data collection functions. *See also* **CONVERSANT VIS**.

VRI	Voice Response Integration
VRU	Voice Response Unit
VuStats	A feature that allows stations equipped with displays to display data items chosen from the BCMS data.
Work mode	An ACD agent’s work state: manual-in, auto-in, after-call-work, or aux (auxiliary).

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RELEASE NOTES

The following CallVisor ASAI G3V4 enhancements are covered in this issue of the *DEFINITY® Communications System Generic 3 CallVisor® ASAI Technical Reference*:

- **ASAI-Accessed Internally Measured Data**

  This feature allows an ASAI adjunct application to access and retrieve Internally Measured Data used to provide VuStats information to display-equipped digital voice stations. The application accesses this data via a new Value Query message that specifies whether agent, split/skill, trunk group, or VDN information is being requested. The new message also specifies a particular observed object of the type (for example, agent or split) specified in the request.

  By providing Internally Measured Data via ASAI, applications can offer enhanced VuStats-like presentations (for example, using graphics, color, or multimedia) on a PC or can further process this data to provide enhanced statistics for better call center management. Also, VuStats-like service can be provided without requiring a digital display-equipped station.

- **Send DTMF Signals**

  This feature is an ASAI-accessed service that, when invoked by a client application, causes the DEFINITY system to send a DTMF sequence on behalf of one party on an active call. The digit sequence to be transmitted is contained in the ASAI service message.

The DTMF tones can be heard by any endpoint connected to the sending talk path. Through such a tone sequence, an adjunct can interact with far-end applications such as automated bank tellers, automated attendants, voice mail systems, various databases, and paging services. An application could provide certain convenience features such as automated entry of passwords or service-access sequences.

- **MultiQuest® Flexible Billing**

  This feature allows an ASAI adjunct to change the rate at which an incoming 900-type call is billed.

- **Redirect Call**

  This feature allows an ASAI adjunct to direct the switch to move an already alerting call away from an extension (at which it is alerting) to another endpoint. Prior to G3V4, such routing was possible only if switch features, rather than an ASAI-provided service, were used (for example, call forwarding or Send All Calls). With this service, an application could, for example, determine, based on call-related information or time of day, etc., whether to answer the call or re-route it to some other number.

  This service does not support removing a call from a queue. Only calls *alerting* at extensions can be manipulated by this service.

- **ASAI-Accessed Integrated Directory Database Service**

  This feature allows an ASAI adjunct to access and retrieve administered name-extension associations stored in the switch (all administered names associated with station extension numbers, trunk groups, VDNs, announcements, logical agents, extensions, ACDs, split/skills, and data extensions). The application accesses this data via a new Value Query message containing the target extension. If a valid Query Message is submitted, the switch responds with a message containing the administered name.

- **Event Report Capabilities**

  Enhancements to Event Reports allow ASAI adjuncts to receive Event Reports when specified events occur at monitored objects. In addition, in the Alerting Event Report, new cause values are provided to specify a reason for redirection.

  The following are descriptions of new/changed Event Reports for G3V4. These reports are only provided if Link Version 2 is active.

  — Agent Login Event Report — This is a new Event Report provided when an agent logs into a domain-controlled ACD group or EAS skill. Notification of manual login events allows adjunct applications to maintain accurate views of current agent login/logout status.

- — Call-Originated Event Report — This is a new Event Report provided when a call is manually or otherwise originated from a domain-monitored station. It also provides the dialed digits to an application.
- — Alerting Event Report — The existing Alerting Event Report has been enhanced to provide a new set of cause code values that map one-to-one to each existing G3V4 reason for redirection that appears on display-equipped stations when a redirected call is offered.

  The new cause codes are provided in an existing Information Element of the Report. An adjunct application can use this new information to provide an enhanced information display (for example, on a computer monitor) to called parties or to otherwise determine how to best handle an incoming call.

In addition to these G3V4 enhancements, a new transport option, ASAI-Ethernet, is available.

CPN/BN on BX.25 links has been removed in G3V4.