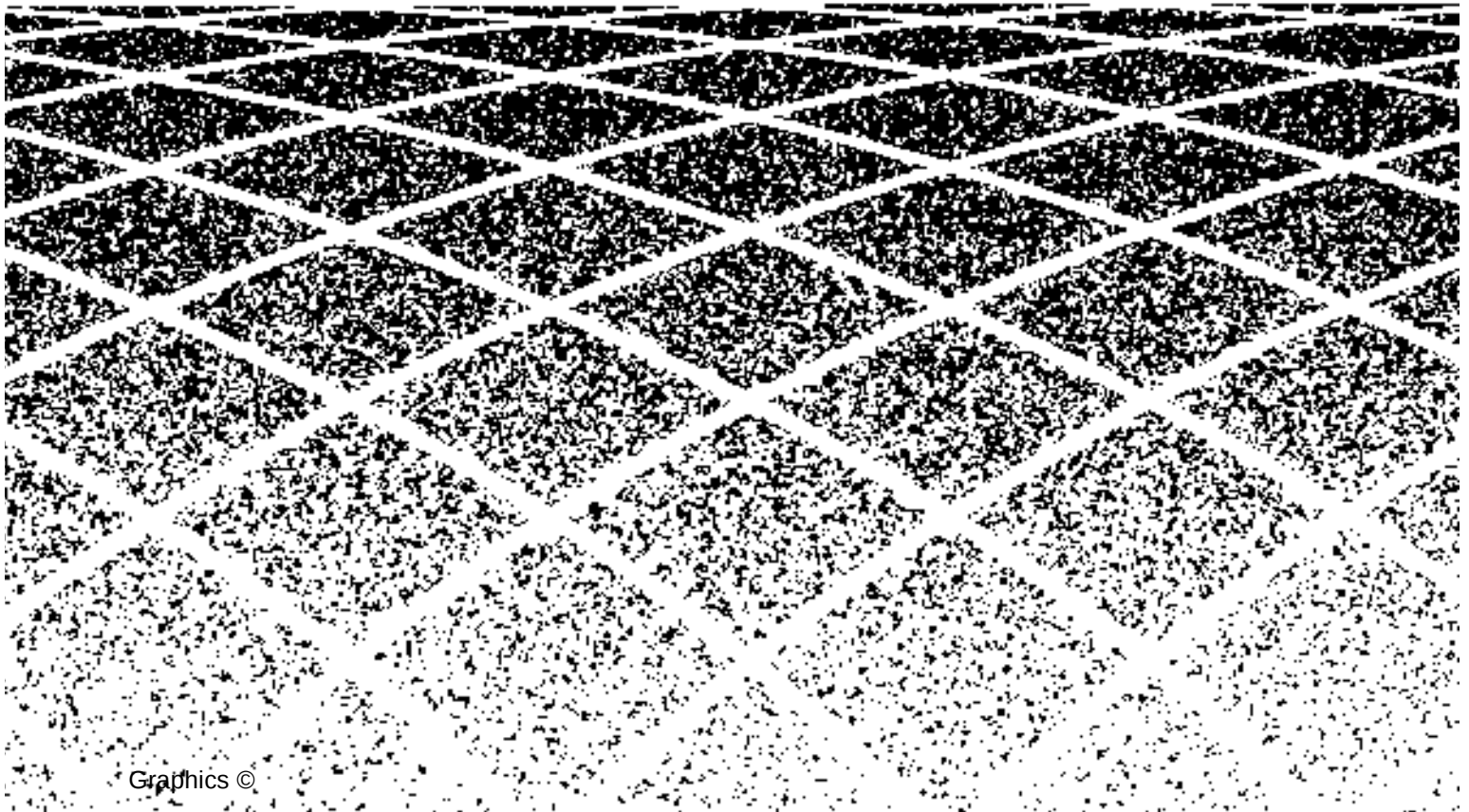




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Conversant VIS Adjunct Switch Application Interface



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Overview of the Adjunct/Switch Application Interface Feature

The AT&T Adjunct/Switch Application Interface (ASAI) is an optional package that may be installed on top of the standard CONVERSANT Voice Information System software. Instructions for installing the Adjunct/Switch Application Interface are provided in Chapter 3, "ASAI Installation".

The AT&T Adjunct/Switch Application Interface provides an Integrated Services Digital Network (ISDN)-based interface between AT&T PBX's and adjunct processors. The VIS ASAI feature supports this application interface for communications with the AT&T DEFINITY Communications System, Generic 3i (hereafter referred to as DEFINITY Generic 3i). This digital signaling interface allows the VIS to monitor and route calls on the DEFINITY Generic 3i switch. When used in conjunction with standard Tip/Ring (T/R) or Line Side T1 (LST1) interfaces, the ASAI interface allows the VIS to monitor and control calls delivered to a T/R or LST1 channel on the VIS.

Access to ASAI capabilities is provided through the Script Builder application generation language. Access to ASAI capabilities at the Script Builder level minimizes the effort required to implement ASAI-based applications. For example, there is no need to receive, parse, and process individual, lower-level ASAI messages. Additional Script Builder actions give the script designer high-level access to ASAI capabilities. These ASAI capabilities can be used to design the following types of applications.

These types of applications are discussed later in this chapter.

- ASAI Voice Response Applications
- Routing Applications
- Data Screen Delivery Applications

These types of applications can run simultaneously on a VIS. This implies that a VIS ASAI system provides co-resident voice response and PBX-to-host gateway capabilities. A single call, for instance, may first be routed by the VIS, handled with a voice response application on the same VIS, and then be monitored by the same VIS as the call is ultimately delivered to a live agent. Furthermore, integration of the voice response and monitoring capabilities allows screens delivered by the host to agents receiving calls to be based on data collected in a voice response script.

ASAI Voice Response Applications

In ASAI voice response applications, incoming calls are routed to the VIS over T/R or LST1 channels which are configured as an Automatic Call Distribution (ACD) split on the DEFINITY Generic 3i switch as depicted in Figure 1-1.

System Monitor - Voice Channels					
Channel	Calls Today	Voice Service	Service Status	Caller Input	Dialed Digits
0	1	appl2	Talking		
1	1	appl4	Collect	8509	
2	3	appl7	Transfer		5312
3	1	appl3	DB	7328	
4	0		*MANOOS		
5	10		Talking		

HELP	PREV-PAGE	NEXTPAGE	PREV-FRM	NEXT-FRM	CANCEL	CMD-MENU	CHG-KEYS
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Figure 1-1. ASAI Voice Response Applications

As a call is delivered to the VIS, the VIS receives ASAI information related to the call. The ASAI feature allows the VIS to recognize the dialed (called) number of an incoming call to a line. This feature is sometimes referred to as Dialed Number Information Service (DNIS). In addition, the ASAI feature allows a service the ability to retrieve the calling party's number. This feature is sometimes referred to as Automatic Number Identification (ANI). This information is used to control which voice application is used for the call. The ASAI information related to the call is also made available to the specific voice application which interacts with the caller. In addition, the call control capabilities of ASAI can be used to transfer the call away from the VIS if the caller needs to speak to a live agent. The following capabilities are therefore provided for ASAI voice response applications:

- **DNIS Service (T/R or LST1 Channel Sharing)** — The DNIS information associated with the incoming call is used to select a particular Script Builder script to service the call. This allows a T/R or LST1 channel to be shared across many applications. Prior to this capability, T/R or LST1 channels were dedicated to specific Script Builder Applications. With channel sharing, the same number of channels can handle more calls while maintaining the same grade of service. Alternatively, the same number of calls can be handled at a higher grade of service.
- **Call Information** — Once the call is answered by the VIS, the ASAI information related to the call such as ANI and DNIS can be retrieved for use in the voice script handling the call.
- **Enhanced Transfer** — The use of ASAI call control capabilities allows the transfer to be faster, quieter (from the caller's perspective), and more reliable. In addition, the DEFINITY Generic 3i ASAI direct agent calling feature can be used to transfer the call to a direct agent. This allows the call to be delivered to a specific agent while maintaining accurate ACD split statistics. Calls placed to specific agents without the direct agent calling feature do not count as ACD calls in calculating and reporting ACD split statistics. Finally, data captured in the voice script can be saved and associated with the transferred call. This enables a host application to deliver to agents data screens which are based on data collected by the voice script which previously serviced the caller.

The availability of ANI within the voice script permits the design of unique voice response applications. Examples include:

- **Locator Service** — A local or host database could be used to determine the closest car dealers, Automatic Transaction Machines, stores, etc.
- **Weather Reports** — Provide a weather report for the callers area.
- **Pay-Per-View** — A cable company could use ANI to automate customer selection of pay-per-view programs.

- Caller Dependent Transfers — The full ten-digit ANI could be used to identify callers and determine where they should be transferred if they need to speak to a live agent. This would be desirable if, for instance, the caller is a preferred customer or is usually handled by a specific agent.
- Geographically-Based Call Transfers — The area code and/or exchange could be used to determine where callers should be transferred if they need to speak to a live agent. This would be desirable if, for instance, agents handle calls from specific geographic regions.

Routing Applications

In routing applications, the VIS is used as a routing server to support the routing capabilities of ASAI and the call vectoring feature on the DEFINITY Generic 3i. However, on receiving a routing request, as depicted in Figure 1-2, a routing application on the VIS receives and responds to call routing requests sent by the DEFINITY Generic 3i.

Note that you are not always required to have the T/R or LST1 lines on the VIS for routing applications. However, T/R or LST1 lines are required to route the call to the VIS agents.

Figure 1-2. ASAI Routing Application **route.pic**

These call routing requests are generated by the DEFINITY Generic 3i when a call is processed by specific call vectors on the DEFINITY Generic 3i.

Information as to where calls should be routed may reside on the VIS in a local database or may be provided by a host to which the VIS is connected. Call routing would typically be based on ANI or call prompting data collected by the DEFINITY Generic 3i.

The use of routing capabilities can significantly improve the efficiency of a call center environment. Examples of routing uses include:

- **Priority Service** — Important or “priority” callers such as large clients can be given priority treatment. A priority caller can be routed to a common agent group but queued at a higher priority so that they are serviced faster. Alternatively, the priority caller can be routed to a specific agent which normally handles their transactions.
- **Call Redirection** — Callers dialing into a particular call center application can be redirected to other call center applications. For example, callers who have delinquent accounts can be redirected to a collections department when they call a sales department.
- **Call Screening** - Fraudulent callers can be disconnected without being connected to an agent so that no network costs are incurred.
- **Geographically Based Service** - In cases where service is provided on a regional basis, callers can be routed to the agent group responsible for their region.

Data Screen Delivery Applications

In data screen delivery applications, a host application delivers a data screen that is specific to a caller or dialed number to an agent at the same time a voice call is delivered to the agent’s telephone. This reduces both the agent time and network time required to service the caller. The reduction in per-call service time translates directly into cost savings. Data screen delivery applications are also known as “coordinated voice/data screen delivery” or “screen pop” applications. A data screen delivery application is depicted in Figure 1-3.

Figure 1-3. Data Screen Delivery Applications gateway.pic

Note that the delivery of data screens is not a function of the VIS itself. A special host application is developed by your company or a third party to perform this function. The VIS acts as a communications gateway between the DEFINITY Generic 3i and the host computer. A monitoring application on the VIS provides the ability to track the status of calls on the DEFINITY Generic 3i. This monitoring application receives information about calls delivered to live agents and forwards this information to the host application. The host application in turn uses this information to deliver a data screen to the agent receiving the call.

The information made available to the host includes which agent receives a particular call and the ASAI information associated with the call such as ANI, DNIS, and any DEFINITY Generic 3i call prompting information collected from the caller. In addition, the call may have been serviced by a VIS voice script and then transferred to a live agent. In this case, information collected in the voice script can be saved and then passed to the host at the time the call is delivered to the agent. Monitoring applications on the VIS can therefore be used to support data screen delivery for three different call flow scenarios as described below:

- **VIS-to-Agent Transfers** — In this call flow scenario, calls are initially delivered to the VIS and then transferred from the VIS to a live agent. Data screens delivered to agents in this scenario can be based on information collected in a voice script in addition to ASAI information such as ANI, DNIS, and call prompting information collected by the DEFINITY Generic 3i.

- Incoming Call Delivered Directly to Agent by ACD — In this call flow scenario, incoming trunk calls are delivered directly to live agents. Here, data screens delivered to agents would be based primarily on ANI, DNIS, and/or call prompting information. Data screens would not be based on data collected in a voice script since a VIS voice script is not used to collect data from the caller.
- Agent-to-Agent Transfers - In this call flow scenario, calls are transferred between live agents. Here, for example, “screening” agents may be used to collect information from the caller and handle simple transactions. The call may subsequently be transferred to “specialized” agents who can handle more complex or detailed transactions. In these scenarios data screens can be based on information keyed in to the host application by live agents. The host application can save data collected and entered by a screening agent and then use this data as the basis for data screens delivered to more specialized agents who may receive the call. Note that the information available for the other two call flow scenarios (that is, ANI, DNIS, call prompting information, and voice script data) is available in this scenario as well. This information may be used in conjunction with data entered by a live agent to provide the basis for data screens.

The VIS-to-agent transfer scenario described previously is supported by using the enhanced transfer capability provided for ASAI voice response applications. The enhanced transfer capability allows data collected in the voice script to be saved and associated with the transferred call. Data saved in this fashion can be included in the call event information passed to the host at the time the transferred call is delivered to an agent.

This ability to save voice script data may be used for many reasons. A voice script can be used to collect a variety of information such as account number, social security number, personal identification number, desired service, etc. In many cases, this type of information is more meaningful and useful than ASAI information such as ANI to the host application and the live agents handling calls. Also, the voice response application might be the primary application providing, for example, an automated banking application. Here, live agents are used only as a backup to answer questions beyond the scope of the voice response application.

The ability to save voice script data with the enhanced transfer capability provides a very useful bridge between voice response and data screen delivery applications. This capability provides true integration (in addition to co-residency) of the voice response and PBX-to-host gateway capabilities offered with the VIS ASAI feature. This mechanism for embedding voice script data in call event information for the transferred call can significantly reduce the complexity of the host application. Without this mechanism, the host application is typically required to associate information from two different physical interfaces (one interface from the voice response unit to receive data collected from the caller and another interface from the monitoring device over which call events are received). Also, the host application is typically required to track and associate multiple events for multiple calls

(the initial incoming call to the voice response unit and the second, transferred call which is delivered to an agent). With the VIS ASAI feature, a single message to the host over a single interface provides all the information needed to deliver a data screen based on data collected in a voice script.

Advantages Using the VIS ASAI Feature

The VIS ASAI feature can greatly improve the operations in your call center environment. The capabilities that this feature offers provide the following benefits to any company that receives customer calls:

- **Enhanced Customer Service**

Caller-dependent and region-dependent treatment for incoming calls is possible in routing and voice response applications. In addition, the direct agent calling feature available with these applications allows calls to be delivered to specific agents while maintaining accurate split measurements. These capabilities help to insure that calls are quickly and reliably directed to the call center resource best suited to handle them. This minimizes the number of transfers a caller experiences and allows callers to be serviced in a rapid, consistent, and personalized fashion and thereby improves customer satisfaction.

In data screen delivery applications, information associated with a given call is available to each agent receiving the call. This reduces customer frustration at having to repeat information to each agent. For example, a caller may be directed initially to a VIS T/R or LST1 channel where the caller is prompted through an automated voice response application. At some point the caller may request to be transferred to a live agent to discuss a topic in more detail. With the VIS ASAI feature, the identity of the caller and additional information collected from the caller by the voice response application is not lost. Pertinent information from the voice response application can be saved and presented in a data screen to the live agent receiving the transferred call, thereby eliminating the need for the customer to repeat information already collected. This reduces call holding time as well as reduces customer frustration. This benefit holds true even when calls are transferred multiple times or are transferred between live agents.

- **Improved Price/Performance**

The co-residency of voice response and PBX-to-host gateway applications greatly improves the price/performance of the VIS ASAI feature over prior and competitive offerings. The VIS ASAI feature eliminates the need for multiple boxes with multiple interfaces to the host computer, thereby simplifying host application development. Access to ASAI capabilities using Script Builder minimizes the effort required to implement the VIS piece of

the overall VIS/host application. In addition, the use of DNIS in voice response applications to enable T/R or LST1 channel sharing means that more calls can be serviced with the same number of VIS channels.

■ **Reduced Cost of Doing Business**

The use of data screen delivery applications reduces the time needed to service calls. This is because the host screen application is ready to provide or accept information at the same time the agent begins to speak with the caller. The reduction in per-call service time translates directly into reduced network costs and reduced agent costs. 800 network charges are lower because calls are shorter. The same number of agents can handle an increase in call volume since per-call service time is reduced. Also, certain calls can be eliminated entirely via the use of routing applications (for example, call screening for the identification of fraudulent calls). In this case, no network costs are incurred for the call and no agent time is wasted on the call.

ASAI Application Planning and Design

2

ASAI Application Planning and Design Overview

This chapter contains information on planning and designing an application that will be used with the AT&T Adjunct/Switch Application Interface (ASAI) feature on the CONVERSANT Voice Information System (VIS).

ASAI Application Planning and Design

Access to ASAI capabilities is provided through the high-level Script Builder application generation language. Subsets of the Notification, Third Party Call Control, and Routing capabilities of ASAI have been integrated into Script Builder for use in ASAI applications. The ASAI feature does not provide access to the Set Value, Value Query, Request Feature, and Third Party Domain Control capabilities of ASAI. The Request Feature capability, however, is used internally by the ASAI feature to log T/R or LST1 channels in and out of an Automatic Call Distribution (ACD) split on the DEFINITY Generic 3i.

The ASAI capabilities supported by the ASAI feature are used to support three classes of applications as follows:

- ASAI Voice Response Applications - The Notification capability of ASAI is used to monitor calls delivered to a VIS T/R or LST1 channel. The ASAI information related to these calls is used to control which voice script is used for the call. ASAI information related to calls is also available to the specific voice scripts which service calls. Refer to Chapter 6, "ASAI Script Builder Actions", for additional information on the A_Callinfo action. The Third Party Call Control capability of ASAI is used when calls need to be transferred away from the VIS.

- **Routing Applications** - The Routing capability of ASAI and DEFINITY Generic 3i call vectoring is used to allow the VIS to act as a routing server. A "routing" script on the VIS receives, processes, and responds to call routing requests sent by the DEFINITY Generic 3i system. Refer to Chapter 6, "ASAI Script Builder Actions" for additional information on the A_Event and A_RouteSel actions.
- **Data Screen Delivery Applications** - The Notification capability of ASAI is used to allow the VIS to monitor calls delivered to ACD agents. A "monitoring" script on the VIS receives information concerning the status of non-T/R or LST1 calls and passes this information to a host for use in presenting data screens to agents receiving calls. Refer to Chapter 6, "ASAI Script Builder Actions" for additional information on the A_Event action.

These three classes of applications can run simultaneously on a VIS. The planning and design considerations for these applications are provided later in this chapter.

The routing and data screen delivery applications are collectively referred to elsewhere in this book as "data" or "data-only" applications. This is to differentiate these types of applications from voice response applications. Similarly, the routing and monitoring scripts used to support these applications are collectively referred to as "data" or "data-only" scripts.

VIS Script Design

Four additional Script Builder actions are provided with the ASAI feature and are used to access ASAI capabilities. These actions are discussed in detail in Chapter 6, "ASAI Script Builder Actions". A brief summary of these actions is provided below for the discussions here.

- **A_Callinfo** — Used within a voice response script to retrieve ASAI information about a call delivered to a T/R or LST1 channel (for example, calling party number (ANI) and called party number (DNIS) for the call). This action therefore provides access to the Notification capability of ASAI for calls delivered to the VIS.
- **A_Event** — Used within routing scripts to receive information about call routing requests sent by the DEFINITY Generic 3i system. This action is also used in monitoring scripts to receive information about calls delivered to an ACD agent. This action therefore serves a dual role by providing access to both the Routing and Notification capabilities of ASAI for non-T/R or LST1 calls.
- **A_RouteSel** — Used within routing scripts to respond to call routing requests previously received via the use of the A_Event action. This action therefore provides access to the Routing capability of ASAI and allows the VIS to send ASAI call routing information to the switch.

- A_Trان — Used within a voice response script to transfer a call away from a T/R or LST1 channel on the VIS. This action makes use of the Third Party Call Control capability of ASAI to effect the transfer.

ASAI Voice Script Design

ASAI voice response applications are designed using the A_Callinfo and A_Transactions within voice response scripts. Other standard Script Builder actions are also used in the voice script to answer the call, greet the caller, collect data, etc. an example of a voice script making use of the A_Callinfo and A_Trان events is included in Appendix A, "Sample Scripts."

The A_Callinfo and A_Trان actions are used only in voice scripts which handle calls delivered to a VIS T/R or LST1 channel. These two actions are not used in routing and monitoring scripts where, in contrast to voice scripts, a call is not present at a VIS T/R or LST1 channel.

For ASAI voice response applications, incoming calls are routed to the VIS over T/R or LST1 channels configured as an ACD split on the DEFINITY Generic 3i system. The Notification capability of ASAI is used by the VIS to monitor this split. As a call is offered to this split, the VIS receives ASAI event reports indicating the status of the call (for example call offered, queued, alerting, and connected event reports). The VIS uses the information contained in these event reports to provide the following capabilities:

- DNIS Service - The Dialed Number Information Service (DNIS) information associated with the incoming call is used to select a particular Script Builder script to service the call. A unique dialed number can be provided for each unique voice response application. Each dialed number would typically be represented by a unique Vector Directory Number (VDN) on the DEFINITY Generic 3i switch. Calls to these different VDN's can be routed to the same VIS split. The DNIS information associated with an incoming call is then used to select a particular application. An administrative screen on the VIS allows the different dialed numbers to be associated with a specific voice response application. This allows T/R or LST1 channels to be shared across many applications. Prior to this capability, channels had to be dedicated to specific Script Builder Applications.
- Call Information - Once the call is answered by the VIS, the ASAI information related to the call can be retrieved for use in the voice script handling the call. In particular, the A_Callinfo action can be used to ANI, DNIS, switch collected user data (call prompting digits), call ID, and incoming trunk group ID if ANI is not available.

A user designing a voice script need not be concerned with processing the individual, lower-level ASAI event reports for incoming calls to the VIS. Rather, special software is provided as part of the ASAI feature. This software processes the event reports and stores the information contained in these event reports on a

per-call basis. The DNIS information associated with a call is used to start a specific voice script on the channel receiving the call. The A_Callinfo action can then be used within the script to retrieve this information and use it in subsequent Script Builder actions.

A subset of the Third Party Call Control capability of ASAI is also supported for ASAI voice response applications. In particular, the A_Tran action uses Third Party Call Control to transfer a call away from the T/R or LST1 channel.

The use of the A_Tran action within a voice response script invokes the Third Party Call Control operations of third party take control, third party hold, third party make call, and third party merge. This sequence of ASAI operations invoked with A_Tran effects a transfer of the incoming T/R or LST1 call to the destination specified with the Destination Number field in A_Tran. Hence, the script designer is not required to program many individual ASAI operations. The use of a single action effects the transfer.

Standard flash transfers are still possible when the ASAI feature is used. The use of A_Tran, however, provides three significant enhancements over existing transfer mechanisms:

- Transfers are faster, quieter (from the caller's perspective), and more reliable since third party call control is used rather than the standard switch-hook flash mechanism.
- The transfer can be completed using direct agent calling. This is done by setting the Destination Number field in A_Tran to the desired agent extension and by setting the Split Extension field to the ACD split logged into by the agent. Direct agent calling allows the transfer to be completed to a specific agent while maintaining accurate ACD split measurements. The DEFINITY Generic 3i direct agent calling feature can only be invoked via ASAI and is therefore not possible via the standard flash transfer mechanism.
- Information captured in the voice script can be saved for subsequent use in a data screen delivery application. Information assigned to the VIS Data field of A_Tran is saved by the VIS even after the voice script terminates. The VIS associates this data with the transferred call and makes this data available in call events passed to the monitoring script which monitors the transferred call.

The third benefit mentioned previously is very useful for data screen delivery applications where the screens delivered to agents are based on data collected by the VIS. Since data collected in a voice script can be saved and is included in call events made available to the monitoring script, the host application is simplified. For instance, a CONNECT event (described later) made available to the monitoring script contains both the extension of the agent receiving the transferred call and the VIS data saved from the voice script which previously serviced the caller.

This single event is then passed to the host, thereby providing all information needed by the host application in a single message.

Routing Script Design

Routing applications make use of the routing capability supported by ASAI and the call vectoring feature on the DEFINITY Generic 3i system. In routing scenarios, calls are not physically delivered to T/R or LST1 channels on the VIS. Instead, incoming calls to the DEFINITY Generic 3i are directed to a vector containing an “adjunct route” step. The adjunct route step causes a “route request” message to be sent to the VIS. The route request message contains information pertaining to the call (for example, ANI). This information is used by the VIS to determine where to route the call.

After the VIS determines where to route the call, a “route select” message is sent back to the DEFINITY Generic 3i system. The route select message contains a destination address provided by the VIS which the DEFINITY Generic 3i uses to further direct the call. In routing scenarios, the VIS may be viewed as a routing server which the DEFINITY Generic 3i calls upon to route calls processed with a routing vector.

Note that, as a result of routing, the call may be directed to a VIS T/R or LST1 split to collect more information from the caller. This would be the case, for example, if the information contained in the route request was not sufficient to identify the caller (for example, ANI not recognized).

Routing applications on the VIS are supported through the use of routing scripts which are designed using the A_Event and A_RouteSel actions. The A_Event action is used to bring information contained in a route request message sent by the DEFINITY Generic 3i system up to the script level. The A_Event action returns a ROUTE REQUEST event when the DEFINITY Generic 3i system has sent such a message. If no route request messages have been sent, the A_Event action waits until one is received. When a ROUTE REQUEST event is made available to the script, it reflects information in an ASAI route request message sent by the DEFINITY Generic 3i system. Note that the A_Event action is also used within monitoring scripts to retrieve other types of events as discussed later.

Once a ROUTE REQUEST event is received in a script and the script determines where the call should be routed, the A_RouteSel action is used to cause a route select message to be passed back to the DEFINITY Generic 3i system. This in turn causes the call to be routed to the desired destination. Unlike voice response scripts, routing scripts are not associated with a particular call. A single routing script handles route requests for many calls. A routing script is designed to receive and process ROUTE REQUEST events which can arrive at any point in time as controlled by vector processing on the DEFINITY Generic 3i system. Hence, the primary difference between routing scripts and voice response scripts

is that, once activated, routing scripts run continuously. Routing scripts, therefore, have the following general structure:

1. An A_Event action to wait for and retrieve a ROUTE REQUEST event from lower-level ASAI software on the VIS. Once the A_Event action retrieves a ROUTE REQUEST event, subsequent actions below are executed.
2. Other standard Script Builder actions which make use of the data made available in the ROUTE REQUEST event to determine where the call should be routed. Examples include read table and get/send host screen actions to retrieve routing information from a local or host database.
3. An A_RouteSel action to pass the routing information (that is, desired destination) from the script to lower-level ASAI software on the VIS. This causes an ASAI route select message containing the routing information to be sent to the DEFINITY Generic 3i system.

Steps 1 through 3 above are repeated by using additional Script Builder steps to create an infinite loop (that is, script labels and Goto actions). A sample routing script is provided in Appendix A, "Sample Scripts."

A routing script may not contain any Script Builder actions which pertain to voice response capabilities (Announce, Prompt and Collect, etc.). A routing script is assigned by using the "RTE" type designation as described in Chapter 4, "ASAI Administration".

A routing script may use any of the information returned in the ROUTE REQUEST event as detailed in Chapter 6, "ASAI Script Builder Actions" to route the call. Examples include the called party number (for example DNIS), calling party number (for example ANI), and switch data (that is, call prompting information). Any one or combination of the data items returned in a ROUTE REQUEST event can be used as the basis for a routing decision.

The call is routed to the destination supplied in the Destination Number field of A_RouteSel. The destination can be on-switch (for example, station, ACD split, or VDN) or off-switch (for example, Direct Distance Dialing number). Also, the call may be routed to a specific agent within an ACD split (direct agent routing). This is done by setting the Destination Number field in A_RouteSel to the desired agent extension and by setting the Split Extension field to the split logged into by the agent. Direct agent routing is the preferred way to route calls to specific ACD agents since direct agent calls are included in the calculations for ACD split statistics (for example, average speed of answer).

Monitoring Script Design

Monitoring scripts on the VIS are used to support data screen delivery applications. The Notification capability of ASAI is used to track the progress of calls that are delivered to agents. A monitoring script on the VIS receives information about these calls and forwards this information to a host application. The host application in turn uses the information to format a data screen presented to agents receiving calls. Note, therefore, that the delivery of data screens is not a function of the VIS itself.

In data screen delivery applications, calls are not physically delivered to a T/R or LST1 channel on the VIS. Rather, calls are delivered to ACD agents on the DEFINITY Generic 3i system. Note, however, that a call may have previously been delivered to a VIS T/R or LST1 channel to collect information from the caller.

A monitoring script is designed using the A_Event action. When used in monitoring scripts, the A_Event action returns the following types of call events:

- CONNECT Event - indicates that a monitored call is being delivered to an agent.
- ABANDON Event - indicates that a monitored call has been abandoned.
- ABANDON events are passed to a script whenever a caller hangs up before being connected to an agent.
- END Event - indicates that a monitored call has ended normally (that is, not abandoned).

Detailed information about the data made available in these events is discussed in Chapter 6, "ASAI Script Builder Actions". The three call event types passed to a monitoring script reflect information contained in ASAI event reports for the call.

Unlike voice response scripts, monitoring scripts are not associated with a particular call. A single monitoring script handles call events for all the calls delivered to a particular domain. A monitoring script is designed to receive and process call events which can arrive at any point in time as determined by how and when calls progress on the DEFINITY Generic 3i system. Hence, the primary difference between monitoring scripts and voice response scripts is that, once activated, monitoring scripts run continuously. Monitoring scripts, therefore, have the following general structure:

1. An A_Event action to wait for and retrieve a call event from lower-level ASAI software on the VIS. Once the A_Event action retrieves a call event, subsequent actions below are executed.
2. Other Script Builder actions used to pass data in the event to a host. Examples include get/send host screen actions to send the data to an IBM host via the standard 3270 interface and a custom external function to pass the data to a custom DIP supporting an asynchronous interface.

Steps 1 and 2 above are repeated by using additional Script Builder steps to create an infinite loop (that is, script labels and Goto actions). A sample monitoring script is provided in Appendix A, "Sample Scripts."

A monitoring script may not contain any Script Builder actions which pertain to voice response capabilities (Announce, Prompt and Collect, etc.). A monitoring script is assigned by using the "VDN", "ACD", or "CTL" type designation as described in Chapter 4, "ASAI Administration".

A monitoring script may pass any combination of the three call event types to a host. In addition, any combination of the data elements returned in a specific call event may be passed to a host. Examples include the called party number (DNIS for example), calling party number (ANI for example), and switch data (call prompting information).

Monitoring scripts on the VIS can be used to support data screen delivery for three different call flow scenarios as described below:

- **VIS-to-Agent Transfers** — In this call flow scenario, calls are initially delivered to the VIS and then transferred from the VIS to a live agent. The transferred call can be monitored with a VDN or ACD type monitoring script if the call is transferred to a monitored VDN or ACD split domain. The transferred call can also be monitored with a CTL type monitoring script allowing the call to be transferred to a non-monitored domain or individual station. If the VIS Data field of A_Tran was used to save voice script data, then this data is made available in the VIS Data field of call events sent to the monitoring script. Hence, data screens delivered to agents in this scenario can be based on information collected in a voice script in addition to ASAI information such as ANI, DNIS, and call prompting information collected by the DEFINITY Generic 3i system. Refer to the information on planning for VIS-to-Agent Transfers in this chapter for additional design considerations.
- **Incoming Call Directly to Agent** — In this call flow scenario, incoming trunk calls are delivered directly to live agents via monitored VDN's or ACD splits. Here, call events are passed to a VDN or ACD type monitoring script and contain only ASAI related information such as ANI, DNIS, and/or call prompting information. Data screens would not be based on data collected in a voice script since a VIS voice script is not used to collect data from the caller. Since the VIS does not service calls in this scenario, no data is present in the VIS Data field of call events.
- **Agent-to-Agent Transfers** — In this call flow scenario, calls are transferred between live agents. Here, for example, "screening" agents may be used to collect information from the caller and handle simple transactions. The call may subsequently be transferred to "specialized" agents which can handle more complex or detailed transactions. In these scenarios data screens can be based on information keyed in to the host application by live agents. The host application can save data collected and entered by a

screening agent and then use this data as the basis for data screens delivered to other, specialized agents which may receive the call. The agent-to-agent transfer can be placed to a monitored domain or to an individual station and is monitored with a VDN or ACD type monitoring script. Note that the call may first have been delivered to the VIS and then transferred to an agent prior to the live agent to live agent transfer. Hence, call events passed to the monitoring script in this scenario can contain the same information available for the other two call flow scenarios. ASAI related information such as ANI, DNIS, and call prompting information and VIS Data can be present in call events. This information may be used in conjunction with live agent entered data to provide the basis for data screens. Refer to the information on planning for Agent-to-Agent Transfers in this chapter for additional design considerations.

VIS-to-Agent Transfers

VIS-to-agent transfers are accomplished by using the A_Trان action within a voice script servicing a caller. The use of A_Trان invokes ASAI Third Party Call Control operations to transfer a call away from the T/R or LST1 channel to which the caller is connected. The caller is transferred to the destination identified in the Destination Number field of the A_Trان action.

The transferred call can be monitored by a monitoring script so that data screen delivery applications can be supported for VIS-to-agent transfers. The transferred call can be monitored in two different ways as follows:

- The call may be transferred to a VDN or ACD split domain monitored by the VIS with a monitoring script. Call events for the transferred call are passed to the script monitoring the domain to which the call is transferred.
- The call may be monitored using a CTL type monitoring script as described Chapter 4, "ASAI Administration". In this case, the call can be transferred to non-monitored domains and individual stations. Here, only call events for calls transferred from the VIS to agents are passed to monitoring scripts. Other direct calls to an ACD split, for example, are not monitored therefore no call events for the direct calls are passed to monitoring scripts.

A combination of the above two monitoring mechanisms may be used on the same VIS. Rules for which monitoring script receives call events when these two mechanisms are combined is discussed in Chapter 4, "ASAI Administration".

In addition to monitoring the transferred call, the application designer has the ability to save data collected in the voice script for subsequent use in the data screen delivery application. This is done by using the VIS Data field of A_Trان. Any data saved in this field when the transfer is initiated from the voice script is presented in call events passed to the monitoring script which monitors the transferred call. The VIS Data field provides twenty characters worth of storage. Note that multiple data items could be stored in this field. A social security number and PIN number,

for example, could be collected in the voice script, concatenated together, and then saved in the VIS Data field. The monitoring script which receives this data in call events could then unbundle the information for use in data screen delivery when the transferred call is delivered to an agent.

The following is a typical call flow for a VIS-to-agent transfer:

1. A call arrives at a T/R or LST1 channel on the VIS. The caller is prompted through a voice response script.
2. The caller decides to speak to a live agent after entering an account number. The voice script transfers the call to a live agent group using the A_Tran action. The account number the caller input is saved by using the VIS Data of A_Tran. The voice script terminates after the transfer is complete and the T/R or LST1 channel is free to handle another call.
3. The transferred call is queued for an available agent. When the call is eventually delivered to an agent, a monitoring script on the VIS receives a CONNECT event for the call. The VIS Data field of this CONNECT event contains the account number previously saved by the voice script. The monitoring script passes the account number along with the connected agent information from the CONNECT event to the host.
4. The host application uses the account number to format a data screen concerning the caller and presents this data screen to the agent receiving the call. The host application does not need to associate multiple calls since all the necessary information is provided in a single CONNECT event for the transferred call.

One CONNECT event is generated for the entire scenario. This is the CONNECT event for the transferred call as it is delivered to the live agent. This CONNECT event contains the VIS Data information in addition to ASAI information related to the *original* call to the VIS. The ANI and DNIS for the original call prior to the transfer, for example, is reported in this CONNECT event. Also, the Other Call ID field contains the call ID of the call originally delivered to the VIS T/R or LST1 channel. Call events for calls to T/R or LST1 channels on the VIS are not passed to monitoring scripts. Also, one END event is generated when the call eventually terminates. As with the CONNECT event, the END event contains data pertinent to the original call. Refer to Appendix B, “Call Flow Examples” for a detailed call flow example.

Additional considerations for VIS-to-agent transfers are as follows:

- In some cases, you may want to collect more data in a voice script than can be stored in the VIS Data field. The recommended method for handling this is to save the data collected by the voice script in the host application. Use A_Callinfo to retrieve the call ID of the call that is delivered to the T/R or LST1 channel. Pass the call ID along with the data to the host from the voice script itself. The host application must buffer the data until

the CONNECT event for the transferred call is received. The call ID in the Other Call ID field of the CONNECT event can be used to correlate the two calls.

- The call may again be transferred after having been serviced by the live agent. In this case, an END event is not reported until all transferring is completed and the call terminates normally. As in the single transfer case, the END event contains information pertinent to the original call. Rules for how subsequent call events are reported is discussed under planning for Agent-to-Agent Transfers in this chapter.
- The discussions on blind and consult transfers, discussed next under planning for Agent-to-Agent Transfers, do not apply to VIS-to-agent transfers completed using the A_Tran action. Also, the delay needed for agent-to-agent transfers discussed later does not apply to VIS-to-agent transfers completed using the A_Tran action.
- Transfers away from the VIS may still be accomplished by using standard flash transfer mechanisms. The use of this type of transfer, however, precludes the ability to use the VIS Data field of A_Tran to save voice script data for later use in data screen delivery applications. Also, the host application must view this type of transfer as an agent-to-agent transfer as discussed next. Hence, the discussions on blind versus consult transfer and the need to introduce delay for blind transfers from the VIS will apply.

Agent-to-Agent Transfers

There are two options for call transfer in an agent-to-agent transfer scenario: blind transfer and consult transfer. These two options differ as to when the screening agent (the agent transferring the call) completes the transfer to the specialized agent (the agent receiving the transferred call) by pressing the Transfer button a second time.

- With a *blind transfer*, the screening agent presses the Transfer button a second time immediately after dialing. The screening agent does not talk to the specialized agent before completing the transfer. In addition, a delay is built into call handling so that the call is distributed to a specialized agent after the screening agent presses the Transfer button the second time.
- With a *consult transfer*, the screening agent waits until the specialized agent answers before pressing the Transfer button a second time. This allows the screening agent to talk to the specialized agent before completing the transfer.

Both of these call-transfer options are described in more detail later. To set up either a blind transfer or a consult transfer, it is important to understand two key concepts of how transferred calls are handled on the DEFINITY Generic 3i system.

The first concept to understand is call monitoring in transfer scenarios. VDN or ACD split domains are monitored by the VIS by assigning monitoring scripts as described in Chapter 4, "ASAI Administration". A call becomes monitored once it enters one of these monitored domains. All domains to which this call may be directed *must* also be monitored by the VIS. Once monitored, therefore, a call remains monitored for the duration of the call even though it may be transferred several times. Once a call becomes monitored, call events are passed to the monitoring script assigned to the domain the call has entered. A CONNECT event, for example, is passed to a monitoring script when a specific agent is selected to receive the call. The screening agent may transfer calls to other monitored VDN's and ACD splits or to individual stations. The original call to the screening agent must be monitored and therefore delivered to the screening agent via a monitored VDN or ACD split.

The second concept to understand is how call ID's are managed in transfer scenarios. The DEFINITY Generic 3i assigns a call ID to each call. The call ID is provided in the Call ID field of call events for the call. In agent-to-agent transfer scenarios there are multiple calls and, therefore, multiple call ID's as described in the following transfer scenario:

1. The original call is delivered to an agent and is assigned a unique call ID. The agent talks with the caller and decides that the call needs to be transferred to another agent.
2. The first press of a Transfer button places the original call on hold and allows another call to be placed from the transferring phone.
3. A second call, temporarily independent of the first call, is placed from the transferring phone. This call is assigned a call ID which is different than that of the original call. If this second call is placed to a monitored domain, the call immediately becomes monitored by the VIS and call events may be passed to a monitoring script. If this second call is placed to an individual station, the call does not become monitored until the transfer is completed as described in Step 4 below.
4. The second press of the Transfer button "merges" the original call which is on hold with the second call and drops the transferring phone from the resultant call.

The VIS is informed about the completed transfer immediately after the merge which occurs in Step 4. It is only after this merge, therefore, that the VIS has the ability to associate the two calls.

With a blind transfer, this merge takes place *before* the merged call is delivered to the second, specialized agent. Hence, with blind transfer calls, the VIS can include information in the CONNECT event for the merged call which relates to the original call. In particular, the call ID of the original call is retained by the VIS and reported in the Other Call ID of call events for the transferred call. This mechanism allows the host application to use call ID's to associate the transferred call with the original call.

With a consult transfer, the merge takes place *after* the second call is delivered to the second, specialized agent. In this case, the original call is still on hold at the first agent's phone when the second call is delivered to the second agent. Hence, for consult transfers, the VIS can only provide information related to the second call in the CONNECT event for the second call. In particular, the call ID of the original call is *not* reported in the Other Call ID field of the CONNECT event for the second call. The host application must use a mechanism other than call ID's to associate the original call with the second call. The alternate mechanism is the Calling Party Number information as discussed later.

Blind Transfer

With a *blind transfer*, the screening agent does not talk to the specialized agent before completing the transfer. With this type of transfer, the VIS retains the call ID of the original call and reports it in the Other Call ID field of call events for the transferred call. Also, other ASAI information such as ANI and DNIS related to the original call is reported in the call events for the transferred call. A typical call flow for blind transfers is described below. In this call flow, Agent 1 is a live agent in a screening split who transfers calls to specialized agents. Agent 2 is a specialized agent that can receive calls via a monitored VDN or ACD split or can be a regular extension. Calls to Agent 1 in the screening split must be delivered via a monitored VDN or ACD split.

1. A call arrives for Agent 1.
2. Agent 1 answers the call and enters pertinent information about the calling customer.
3. Agent 1 transfers the call to Agent 2. This is done by pressing the transfer button, dialing the VDN, split, or individual extension and pressing the transfer button again.
4. Agent 1 is finished with the call.
5. The host application uses call ID information reported in CONNECT events to determine which data to display on Agent 2's data-terminal screen. The call ID from the Call ID field of the CONNECT event for the original call matches the call ID provided in the Other Call ID field of the CONNECT event for the transferred call.

Two CONNECT events are passed to monitoring scripts for the entire scenario, that is, one for the original call to the screening agent and one for the transfer to the specialized agent. One END event is generated when the call eventually terminates. Refer to Appendix B, "Call Flow Examples" for detailed call flow examples which include complete descriptions of call flows and call event contents.

The persons responsible for administration and application development must thoroughly understand the call flow described previously. In addition, note the following:

- The domain receiving the original call and any domains receiving the transferred call must be monitored.
- In call-center operations that use blind transfer, the host application may tag current call data by call ID. The call ID allows the application to determine which data is associated with the call as the call is transferred to a monitored domain or station.
- Calls can be transferred either to a monitored domain or to a station. For a blind transfer to a monitored domain, the following must be considered:
 - The agent must complete the transfer immediately after initiating it by pressing the Transfer button a second time.
 - A delay must be built into the flow of the transfer so that the completion of the transfer can be recognized by the communications system before the receiving agent is selected for the call. You can create this built-in delay by transferring calls to a VDN. This VDN is associated with a vector which has a “wait” step in it. The vector would then direct the call to the desired split with a “route to” or “queue to” step.

For blind transfer to a station, the following must be considered:

- When an agent in a monitored domain completes a transfer to a station rather than to a split, a CONNECT event is passed to a monitoring script. The agent must initiate and complete the transfer by pressing the Transfer button a second time in order for the CONNECT event to be passed to the script. The CONNECT event therefore only becomes available to the host application when the Transfer button is pushed the second time.
- If for some reason calls are transferred to non-monitored domains, unexpected operation can result. When the call placed by Agent 1 is not initially monitored, the VIS assumes that a transfer to a station is taking place. Hence, two CONNECT events for the transferred call would be generated. One CONNECT event would be generated when the transfer is completed by Agent 1 and another would be generated when the call is actually delivered to Agent 2. Also, the Connected Party Number field of the first CONNECT event for the transferred call would identify the ACD split or VDN extension dialed by Agent 1. This is as opposed to identifying the extension of Agent 2. Note that the Connected Party Number field of the second CONNECT event for the transferred call would identify the extension of Agent 2.
- The END event that is reported for the transferred call contains information pertinent to the original call. For example, the original ANI for the caller is reported in the Calling Party Number field and the call ID for the original call is reported in the Other Call ID field. Also, an END event is reported for a call only when the call ultimately terminates. An END event is not reported when a call is transferred.

- The call may again be transferred after having been serviced by Agent 2. In this case, an END event is not reported until all transferring is completed and the call terminates normally. As in the single transfer case, the END event contains information pertinent to the original call. Rules for how subsequent CONNECT events are reported are as described in this chapter and depend on whether the call is transferred to a monitored domain or to a station and whether consult or blind transfer operation is used.

Consult Transfer

With a *consult transfer*, the screening agent talks to the specialized agent before completing the transfer. With this type of transfer, the call ID for the original call is not retained by the VIS and reported in the Other Call ID field of call events for the transferred call.

A typical call flow for consult transfers is described below. In this call flow, Agent 1 is a live agent in a screening split who transfers calls to specialized agents. Agent 2 is a specialized agent that can receive calls via a monitored VDN or ACD split or via an individual station. Calls to Agent 1 in the screening split must be delivered via a monitored VDN or ACD split.

1. A call arrives for Agent 1.
2. Agent 1 answers the call and enters pertinent information about the caller.
3. Agent 1 presses the Transfer button.
4. Agent 1 dials the extension of the monitored domain or station to which the call will be transferred.
5. Agent 1 waits for Agent 2 to answer.
6. Agent 1 and Agent 2 consult privately about the caller.
7. Agent 1 presses the Transfer button again to complete the transfer.
8. Agent 1 is finished with the call.
9. The host application uses calling party information to determine which data to display on Agent 2's data-terminal screen. The extension for Agent 1 is reported in the Calling Party Number field of the CONNECT event for the second call.

Two CONNECT events are passed to monitoring scripts for the entire scenario one for the original call to the screening agent and one for the call to the specialized agent. One END event is generated when the call eventually terminates.

Refer to Appendix B, "Call Flow Examples" for detailed call flow examples which include complete descriptions of call flows and call event contents.

The persons responsible for administration and application development must thoroughly understand the call flow described previously. In addition, note the following:

- With a consult transfer, Agent 1 and Agent 2 generally both view the call data in a private consultation while the caller is on soft hold.
- Calls can be transferred either to a monitored domain or to an individual station. For a consult transfer to a monitored domain, the following must be considered:
 - When Agent 2 is selected to receive the call from Agent 1, a CONNECT event is made available to a monitoring script. Since Agent 1 stays on the line until Agent 2 answers, the two calls have not yet have been merged. This implies that the CONNECT event for the second call does not contain information pertinent to the first call. The Other Call ID field for the second CONNECT event, for example, is NULL and does not contain the call ID of the first call. Also, for example, the Calling Party Number field contains the extension for Agent 1 and not the ANI for the caller. This is because the second call is viewed as a new, direct call to Agent 2 from Agent 1. The VIS cannot assume that the two calls will eventually be merged since, in some cases, they may not be merged. Hence, the two calls cannot be correlated by using call ID's from CONNECT events.
 - In this case, the Calling Party Number field of the second CONNECT event should be used to correlate the two calls. This field contains the extension for Agent 1. The host application can assume that Agent 1 is performing a consult transfer. The host application could then retrieve the appropriate data from Agent 1's data-terminal screen and deliver it to Agent 2's data-terminal screen. After the two agents consult, Agent 1 can complete the transfer by pressing the Transfer button a second time. No additional CONNECT event is passed to a monitoring script when the transfer is completed.

For consult transfer to a station, the following must be considered:

- A CONNECT event for the second call is passed to a monitoring script only after the transfer is completed when the Transfer button is pressed the second time by Agent 1. This means that when Agent 1 and Agent 2 are talking privately, the host application will not have been notified about the second call to Agent 2. This is because the second call is placed to a station and not to a monitored domain. The VIS, therefore, does not receive events for the second call until the two calls are merged. The host application can be programmed to allow the receiving station to query for the data. After the transfer is complete, a CONNECT event is passed to a monitoring script. This CONNECT event contains information pertinent to the first call. The Other Call ID field of this CONNECT event, for example, con-

tains the call ID of the original call delivered to Agent 1. Also, for example, the Calling Party Number field of this CONNECT event contains the ANI of the original caller.

- If for some reason calls are transferred to non-monitored domains, unexpected operation can result. When the call to Agent 2 from Agent 1 is not initially monitored, the VIS assumes that a transfer to a station is taking place. Hence, the Connected Party Number field of the CONNECT event generated when the transfer is completed by Agent 1 would identify the ACD split or VDN extension dialed by Agent 1. This is as opposed to identifying the extension of Agent 2.
- The END event that is reported for the transferred call contains information pertinent to the original call. For example, the original ANI for the caller is reported in the Calling Party Number field and the call ID for the original call is reported in the Other Call ID field. This is true even though the second CONNECT event for consult transfers to monitored domains does not contain information pertinent to the original call. This is because the END event is reported after consult transfers to monitored domains are completed (that is, after the two calls are merged and can be associated by the internal software on the VIS). Also, an END event is reported for a call only when the call ultimately terminates (that is, an END event is not reported when a call is transferred). These properties for END events allow the host application to consistently use the Other Call ID field of END events to identify when and which calls have left the DEFINITY Generic 3i entirely.
- The call may again be transferred after having been serviced by Agent 2. In this case, an END event is not reported until all transferring is completed and the call terminates normally. As in the single transfer case, the END event contains information pertinent to the original call. Rules for how subsequent CONNECT events are reported are as described in this chapter and depend on whether the call is transferred to a monitored domain or to a station and whether consult or blind transfer operation is used.

Host Application Planning and Design

In certain call center environments, the VIS ASAI system is integrated with a host computer. As discussed previously, you must provide or obtain the host software application that works with the VIS ASAI system. This host software application is not part of the VIS ASAI product. The host application can use the information it receives from the VIS ASAI system to do certain functions such as display call information on agent screens or route calls. The host application may also be called upon to provide the basis for an automated voice response application.

In some cases, particularly for voice response applications, the VIS ASAI system integrates well with an embedded application and hence no changes are required. For routing and data screen delivery applications, however, an existing application will most likely need to be modified to accommodate new functionality.

You may have several options for providing this host application. For example, you can develop your own application or modify an existing application to work with the VIS ASAI system. This is typically done by the company's data-processing or information-systems department. Alternatively, you can purchase a third-party software vendor application that is designed and developed to work with the VIS ASAI system.

Application development may require significant planning and coordination between different organizations within your company. The telecommunications, call-center operations, and data-processing organizations are all typically involved in the planning process. Schedules for application development or customization must be coordinated closely with plans to implement the VIS ASAI system, Integrated Services Digital Network (ISDN) services, and any additional communications system ACD features.

The voice response, routing, and data screen delivery applications enabled by a VIS ASAI system can all potentially make use of ANI information delivered by the network. The use of ANI generates several considerations.

- You should allow for the possibility that the same caller will call from different phone numbers. The same person, for example, might sometimes call from home and sometimes call from the office. The same database record should be used in both cases. Calls generated from a private branch exchange (PBX) will likely have more than one ANI assignment, based on the different trunk groups used to generate the call and the fact that individual trunk circuits sometimes carry different ANI identities.
- You should allow for situations when ANI information is not delivered for a call. In voice response applications, the voice script should provide some sort of default call handling for cases where no ANI is available. In routing applications, the caller could be routed to a VIS T/R or LST1 split so that additional information can be collected. In data screen delivery applications, an agent can ask the caller for this information.

- You may want to write an ANI learning module to automatically associate new ANI information with existing customer records. Agents and voice response scripts can verify ANI information passed by the DEFINITY Generic 3i to the VIS.
- You should allow for situations where a single ANI is associated with multiple calling customers. More than one customer, for example, can call from the same PBX. Examples of how to handle such situations include bringing up a menu from which the agent can choose the appropriate customer and switching to traditional methods for bringing up customer data.

ASAI Voice Response Application Considerations

- Voice response applications can make use of direct agent calling. Calls can be transferred to specific agents within ACD splits after being serviced by a voice response script. In this case, your database must maintain the ACD split extensions that agents are logged into as well as the extensions for the agents themselves.
- If your voice response applications involves transfers to live agents, refer to the information on planning for VIS-to-Agent Transfers in this chapter for additional design considerations.

Routing Application Considerations

- Unlike data screen delivery applications, routing applications make use of the host application in an “inquiry/response” fashion. This implies that the addition of a VIS ASAI routing application to your call center may have little or no impact on the high-level operation of the application. The most significant change to the host application will likely be the information stored in the database. Information as to how calls should be routed must be added to the database if it is not already present. An example is ANI-to-agent and/or ACD split mappings. If feasible, consider using a local VIS database to store routing information.
- Routing applications can make use of direct agent calling. Calls can be routed to specific agents within ACD splits. In this case, your database must maintain the ACD split extensions that agents are logged into as well as the extensions for the agents themselves.

Data Screen Delivery Application Considerations

- Prior to the use of data screen delivery applications, a host application typically waits for input from agents before performing an operation. Thus, the agent's input generally controls the application. With data screen delivery applications, a new input to the application is provided. While this input enables agents to work more quickly, it means that the host application must be modified. In particular, the call events provided by a monitoring script on the VIS must be used by the host to drive the screens on agent's terminals. The interface to the VIS serves as a "control" link while the interface to the agent operates traditionally as an "inquiry/response" link. The interactions between these two properties of the application must be considered carefully.

Suppose, for example, that a call arrives for an agent before that agent has finished entering data from the previous call. This scenario can be handled in one of two ways:

- Agents can be trained to use Aux Work or After Call Work feature buttons on their phones to make themselves unavailable for calls until they have finished entering data from the previous call.
 - There is typically a point in the application's sequence of operations (for example, base transaction screen) where the agent is waiting for a new call and begins interaction with the application. The application could look for information from the VIS only at this point. The agent's phone will alert the agent to the new call, and the agent can quickly finish work on the previous call. You may wish to provide a quick way for the agent to move to this place in the interactions with the application.
- In data screen delivery applications, phone extensions are used to identify agents receiving calls. The host application must therefore be able to associate extensions with particular data terminals. Three possible ways of doing this are:
 - The agent can be queried for the phone extension when the application is started. This is the most flexible arrangement and handles the situation where data terminals and terminal ID's are not permanently associated with the same phone. Agents do make mistakes in providing the phone extension to the system. You should plan for these occasional mistakes and make sure agents understand what must be done to properly use the system. Discuss this issue with the person responsible for the company's agent operations.
 - If an agent is always assigned to the same work position and hence extension, the extension information could be added to an agent profile.

- If the relationship between data terminals, terminal ID's, and telephones is relatively stable, administration of the host application can maintain a fixed mapping between phones and terminals.
- The agent screen application should be able to operate even if the VIS is not delivering call events. If call information is not being delivered, the appropriate person or the application itself should notify agents that there is a problem and that they should operate in manual mode. The DEFINITY Generic 3i continues to deliver calls to agents even if the ASAI link to the VIS is down.
- If your call center application involves data screen delivery for VIS-to-agent transfers, refer to the information on planning for VIS-to-Agent Transfers in this chapter for additional design considerations.
- If your call center application involves data screen delivery for agent-to-agent transfers, refer to the information on planning for Agent-to-Agent Transfers in this chapter for additional design considerations.
- Your application should be able to accommodate cases where there are multiple CONNECT events received for the same call. This can occur, for example, in cases where direct agent calling is used. A call may first ring at the initial agent's phone and then at the phone of a covering agent if the call is not answered by the initial agent. In this case, two CONNECT events are sent to a monitoring script when CONNECT events are triggered on an ASAI alerting event report.
- Your application should be able to accommodate cases where the connected party identified in call events is not a known ACD agent. Depending on switch administration and the design of call vectors, calls can be redirected to domains (VDNs or ACD splits) other than the domain to which the call is originally offered. If calls cover or are redirected from a live agent split to an AUDIX split, for example, call events can identify an AUDIX channel extension as the connected party.

Communications System Planning

Communication system planning involves defining what changes must be made to your company's communications system software configuration and ACD environment to support the planned applications. The following is a list of items that should be considered when planning for the changes to the communications system.

- A DEFINITY Generic 3i system with ASAI software is required to implement VIS ASAI applications.
- Call vectoring is strongly recommended for use in implementing all VIS ASAI applications. This is especially true for data screen delivery applications which involve agent-to-agent transfers or DNIS service and voice response applications which make use of DNIS service.
- Call vectoring is mandatory for routing applications. Call vectoring is also mandatory for data screen delivery applications which make use of call prompting information. Note that the call prompting capability of vectoring is an additional, optional feature over and above the optional call vectoring feature.
- If feasible, you may wish to aggregate agents currently in multiple splits into a single split. This minimizes the number of domains that are monitored by the VIS and allows agents to be used more efficiently. Since DNIS is available in call events, you can have a single split of agents handle several applications. The host application can use DNIS to provide information screens that tell agents how to answer and handle calls.
- You must plan your call flows carefully if multiple ASAI adjuncts will be used with the same DEFINITY Generic 3i system. Once a call becomes monitored by a particular VIS, the call cannot be redirected or transferred to a domain monitored by another VIS or ASAI adjunct. This is a consideration primarily for data screen delivery applications. For example, if you have agent-to-agent transfers for data screen delivery applications, agents must restrict transfers to domains monitored by the same VIS that monitors calls delivered to them. Also, for example, you may have VIS-to-agent transfers to support data screen delivery based on VIS collected data. In this case, multiple VIS systems should be configured to "front end" mutually exclusive sets of live agents. These considerations do not apply if only one VIS ASAI system is used and it is the only ASAI adjunct.
- If ANI is needed for an application, you must provision an ISDN Primary Rate Interface (PRI) in order to receive ANI from the network. Make sure ISDN service is available in your area when needed.

Call Center Operations Planning

The persons responsible for call center or customer service operations should plan for the changes that occur in those operations when the VIS ASAI system is added to the call center. In particular, the use of automated voice response applications reduces the volume of calls handled by live agents. In addition, the use of routing and data screen delivery applications can make the call center even more efficient by reducing the amount of time required to service those calls that are handled by live agents. This should reduce the amount of time that customer calls are in queue thereby increasing the productivity of agents and reducing the use of trunk facilities. You can take advantage of these benefits in one of two ways:

- You can improve customer service by keeping the same number of agents and reducing the average speed of answer (ASA). ASA is the average time, computed by Call Management System (CMS) or Basic Call Management System (BCMS), that callers wait in queue before connecting to an agent.
- You can save money by reducing the number of agents and keeping approximately the same ASA.

Efficient agent operation is the key to realizing the improvements the VIS ASAI system can provide. Specific agent tasks may change when a VIS ASAI application such as data screen delivery is added to the call center. You should determine what agent training is needed before the new service begins. Agents should be trained on what new information will appear on their data-terminal screens and how to use that information to interact with calling customers. Before implementing a data screen delivery application with the entire agent population, conduct a trial to compare old call center operations with the new call center operations using a data screen delivery application. Benefits of the application should be explained so that agents can take advantage of them.

If data screen delivery is performed for agent-to-agent transfers, carefully read the information on planning for Agent-to-Agent Transfers in this chapter. Agents must be trained to perform transfers properly so that the desired call events are passed to the host application. More specifically, for blind transfers, agents must transfer calls as follows:

1. Place the original call on hold by hitting the Transfer button once. This also causes a new call appearance to become active (dial tone is heard on this call appearance).
2. Dial the desired extension while hearing dial tone on the new, active call appearance.
3. Immediately press the Transfer button again after dialing the desired extension to complete the transfer.

In *consult* transfer scenarios, the agent may wait to talk to the second agent before completing the transfer. In consult transfer scenarios, however, the agent

must make sure that the original call is on “transfer” hold before completing the transfer. A call is said to be on transfer hold when the call was placed on hold by hitting the Transfer button. This is as opposed to “regular” hold where the call is placed on hold by hitting the Hold button.

For example, the agent may decide to return to the original caller before completing the transfer (for example to say, “please wait while I transfer you to Bill who can handle your question”). When the agent returns to the original caller, the original caller is no longer on transfer hold. The agent must be sure to place the original call on transfer hold (not regular hold) before completing the transfer.

The following procedure should be followed for consult transfer situations where the screening agent wants to go back and talk to the original caller before completing the transfer. In this procedure, Agent 1 is the screening agent who receives the original call from the calling customer. Agent 2 is the specialized agent who receives the transferred call. Although this procedure may seem cumbersome initially, it is the most natural set of steps to take in consult transfer scenarios where the screening agent wants to announce the transfer to the original caller after having talked to the specialized agent. This procedure also insures that the VIS can properly identify the original call when the two calls are merged. If agents do not follow this procedure, inaccurate call events are reported to the host application.

1. Agent 1 places the call to Agent 2 on regular hold by hitting the Hold button while the call to Agent 2 is still the active call.
2. Agent 1 returns to the original caller by pressing the call appearance for the original call. This makes the original call active once again. Agent 1 may now talk to the original caller.
3. After talking to the original caller for the second time, Agent 1 places the original caller on transfer hold again by pressing the Transfer button again. This is the second time Agent 1 has pressed the Transfer button. This causes a third, as yet unused, call appearance to become active (dial tone is heard on this call appearance but this call appearance is not used for anything - go to the next step and ignore the dial tone).
4. Make the call to Agent 2, which is currently on regular hold, the active call by pressing the call appearance for this call. At this point Agent 1 and Agent 2 are connected again and Agent 1 can inform Agent 2 that the transfer is about to be completed.
5. Agent 1 completes the transfer by hitting the Transfer button again. This is the third time Agent 1 has pressed the Transfer button.

Installation Overview

This chapter contains information on installation and setup procedures for the CONVERSANT Voice Information System (VIS) Adjunct/Switch Application Interface (ASAI) software package. ASAI-specific installation processes are discussed in this chapter. Refer to Chapter 2, "Installing The Base System Software" in *CONVERSANT VIS Software Installation*, 585-350-111 for additional information on general voice system installation procedures.

Prerequisites for ASAI Installation

To install the Adjunct/Switch Application Interface feature on the CONVERSANT Voice Information System (VIS) Version 3.1, you need the following:

- One or more Tip/Ring cards (IVP4, IVP6, or a VRS6) with Signal Processor installed
- OR
- one or more T1 cards (AYC11 or AYC3B) with Signal Processor installed
- Network Support Utilities (1.2) Version 2.0
- One ISDN PC Interface (IPCI) card
- The Adjunct/Switch Application Interface feature package
- Configurator program output

For information on installing the VIS software and hardware, refer to the CONVERSANT Voice Information System Software Installation and Upgrade and the hardware installation and upgrade book for your platform.



NOTE:

Confirm that Adjunct/Switch Application Interface is installed and available on the switch you are using. Refer to the DEFINITY Communications System Generic 1 and Generic 3i Installation and Test Manual , 555-230-104.

ASAI Hardware Architecture

The VIS is designed to operate with the DEFINITY Generic 3i. The IPCI card must be installed on the VIS. The ISDN Line Circuit Pack, TN556, must be installed on the DEFINITY Generic 3i. For information on the TN556 refer to the DEFINITY Communications System Generic 1 and Generic 3i System Description, 555-230-200 . Refer also to the DEFINITY Generic 1 and Generic 3i Wiring Manual, 555-204-111, Issue 2 .

To support the ASAI capability, the VIS must be connected, via a point-to-point ISDN Basic Rate Interface (BRI), to the DEFINITY Generic 3i. The VIS (T/R or LST1) lines which need to access ASAI capabilities must be configured as members of an Automatic Call Distribution (ACD) split of the PBX. The IPCI card supports the BRI D-channel interface from the switch. One ASAI link per VIS is supported.

A typical VIS and DEFINITY Generic 3i configuration is displayed in Figure 3-1.

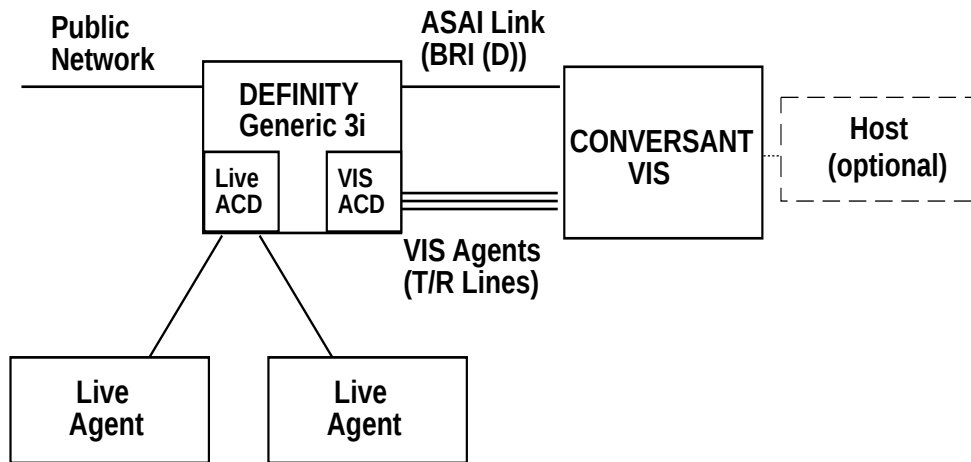


Figure 3-1. Typical Hardware Architecture

Installing ASAI Hardware

Prior to installation, you should run the configurator program. For information on running the configurator program, refer to Chapter 3, "Allocating System Resources" in the hardware installation and upgrade book for your platform. Referring to the output from the `show_config` command and Figure 3-3, set the Base RAM address selection switches on the IPCI card. Figure 3-2 shows the location of the Base RAM address location switches. Figure 3-3 shows the various switch settings for the Base RAM selection switches.

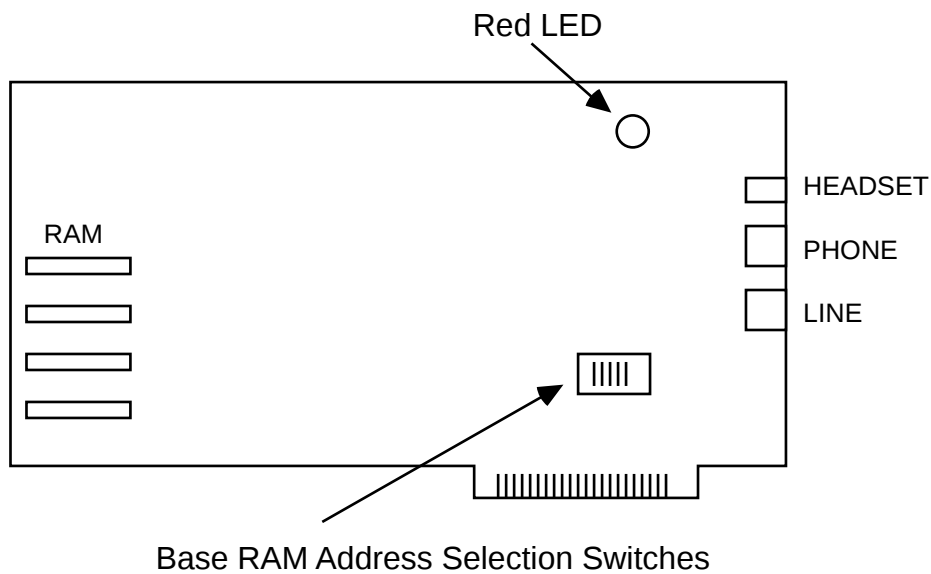


Figure 3-2. IPCI Card

Base Address	Set These Switches			
	1	2	3	4
0c8000h	ON	OFF	OFF	ON
0cc000h	OFF	OFF	OFF	ON
0d0000h	ON	ON	ON	OFF
0d4000h	OFF	ON	ON	OFF
0d8000h	ON	OFF	ON	OFF
0dc000h	OFF	OFF	ON	OFF

Figure 3-3. IPCI Switch Settings

For information on installing the IPCI Card in the system, refer to Chapter 5, "Installing Circuit Cards" in the hardware installation and upgrade book for your platform.

Figure 3-4 shows a typical wiring architecture for the ASAI link.

Note that you must connect the AT&T 440A4 8-pin terminating resistor (or equivalent) to the LINE connector of the IPCI Card using the provided DW8 cable. Use the other DW8 cable to connect from the connecting block to the terminating resistor.



CAUTION:

Total cable length from the DEFINITY Generic 3i system to the VIS should not exceed 1900 feet.

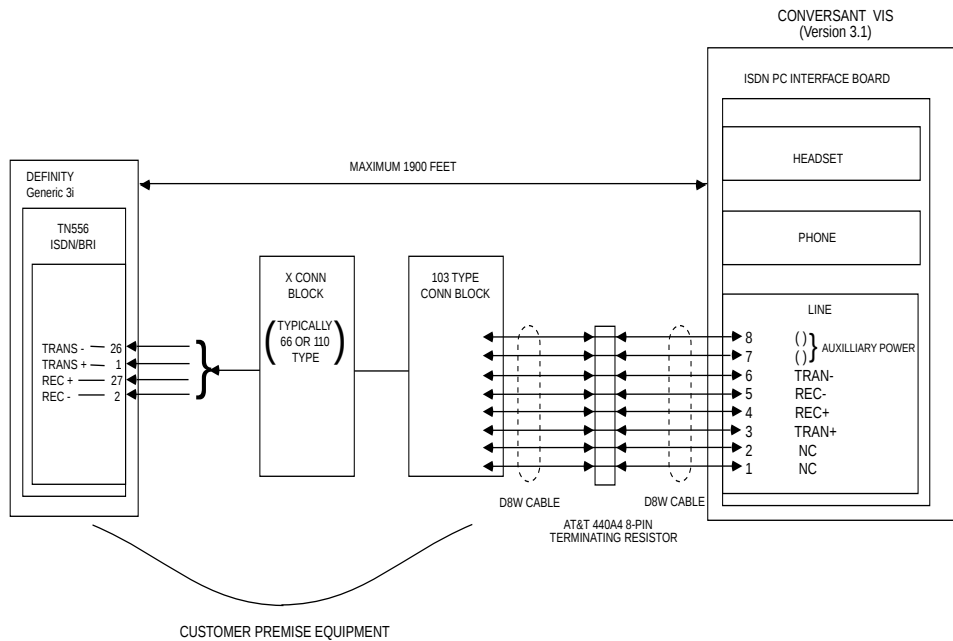


Figure 3-4. Typical Wiring for ASAI Link **wiring.ps**

Installing ASAI Software

The VIS ASAI software package is contained on high-density 3.5" floppy disks. Use the following procedures to install the software package.

6. Log in to the VIS as root.
7. Type **installpkg** at the UNIX system prompt.
8. Insert the floppy disk labeled "Adjunct/Switch Application Interface Library" into the disk drive and press **(ENTER)**.
An **Install in progress** message appears on the screen.
9. The package prompts you for the number of IPCI cards installed on the system. Type **1** and then press **(ENTER)**. Currently only one (1) link is supported per VIS.

10. Next, the package prompts you for the Interrupt Vector Number (IVN) of the IPCI card. Refer to the output of the configuration program to determine what interrupt number to use. Type in the number you have selected and then press **(ENTER)**. If you wish to cancel the software installation process at this time, you may type **q** to abort installation.
11. You are now prompted to enter the Memory Address of the card. Refer to the information under "RAMADDR" from the output of the configuration program for this setting for the IPCI card.
12. The UNIX system will now be rebuilt. This takes approximately 2 minutes to complete.
13. When prompted, remove the disk from the drive.
14. Press **(ENTER)** to shutdown the system.
15. When prompted, reboot the system. While holding down **(CONTROL)** and **(ALT)** press and release **(DELETE)** on the numeric keypad, then release **(CONTROL)** and **(ALT)**.
16. Log in to the system as root.
17. Type **installpkg** at the UNIX system prompt.
18. Insert the floppy disk labeled "CONVERSANT ASAI Package Version 3.1" and press **(ENTER)**.
19. Remove the floppy disk when prompted, and load the remaining floppies into the disk drive when directed to do so. Press **(ENTER)**.
20. If the Voice System is running, the system asks you whether you wish to stop the voice system. Type **y** and then press **(ENTER)** to stop the voice system.
21. A message appears stating that the installation of the ASAI package is now complete.
22. A message appears asking you if you wish to restart the voice system. Type **y** and then press **(ENTER)** if you wish to start the voice system at this time. Type **n** if you do not wish to start the voice system at this time (that is, if you have other software packages to load).

You have now completed loading the ASAI software. Refer to Chapter 4, "ASAI Administration" to begin work with the various ASAI features and capabilities.

Removing the ASAI Software

To remove the ASAI software, refer to the following procedure:

⇒ NOTE:

The ASAI feature package software must be removed before the ASAI Library package software.

1. Before you remove the ASAI feature package software or the ASAI library package software, you will need to stop the voice system. This may be done by using either the System Control screen or, from the UNIX system command line, by using the **stop_vs** command. Refer to Chapter 5, "Configuration Management" in *CONVERSANT Voice Information System Operations* for information on stopping the voice system.
2. From the system prompt, type **removepkg** . The numbered list of installed packages is displayed.
3. Select the number associated with the ASAI package.
4. When the prompt is returned, the ASAI software package has been removed.

This chapter describes the administrative screens that are provided with the ASAI feature package. For a step-by-step description of a typical voice-response application involving a VIS ACD and VIS agents, refer to Chapter 5, "Administering ASAI".

ASAI Administration Overview

⇒ NOTE:

The ASAI feature uses a menu-driven user interface. Refer to Chapter 1, "User Interface" of the *CONVERSANT VIS Version 4.0 Operations*, 585-350-703, for a general description of the VIS user interface. The ASAI link must be administered on the DEFINITY Generic 3i. Refer to the *DEFINITY Communications System Generic 3i Implementation*, 555-230-650, for additional information.

After you have logged into the VIS, the CONVERSANT VIS Version 4.0 screen should appear. If the CONVERSANT VIS Version 4.0 screen does not appear, type **cvis_mainmenu** at the system prompt to display the CONVERSANT VIS Version 4.0 screen. From this screen, use the arrow keys to highlight "Voice System Administration", then press **ENTER**. The Voice System Administration screen appears in Figure 4-1.

Voice System Administration
Application Package Administration
Configuration Management
Feature Packages
Reports
Script Builder Applications
Switch Interfaces
System Monitor

Highlight an item and press ENTER.

HELP			PREV-FRM	NEXT-FRM	CANCEL	CMD-MENU	CHG-KEYS
------	--	--	----------	----------	--------	----------	----------

Figure 4-1. Voice System Administration Screen

From the Voice System Administration screen, highlight “Feature Packages” and press **ENTER** to open the Feature Packages screen (Figure 4-2).

Feature Packages
ASAI Administration

Select a feature package and press Enter

HELP			PREV-FRM	NEXT-FRM	CANCEL	CMD-MENU	CHG-KEYS
------	--	--	----------	----------	--------	----------	----------

Figure 4-2. Feature Packages Screen

To access ASAI administration from the Feature Packages screen, highlight “ASAI Administration”, then press **ENTER** to display the ASAI Administration screen (Figure 4-3).

ASAI Administration
Channel Administration Diagnose IPCI Board Domain Administration Initialize IPCI Board Parameter Administration Show ASAI Software Version Show Status of ASAI Link Take IPCI Board Off-line

Please highlight the item you want and press Enter

HELP					CANCEL		CHG-KEYS
------	--	--	--	--	--------	--	----------

Figure 4-3. ASAI Administration Screen

The ASAI Administration screen enables you to access the following administrative features associated with ASAI:

- Channel Administration
- Diagnose ISDN PC Interface (IPCI) Board
- Domain Administration
- Initialize IPCI Board
- ASAI Parameters
- Show ASAI Software Version
- Show Status of ASAI Link
- Take IPCI Board Off-line

Channel Administration

The Channel Administration screen provides one entry for each T/R or LST1 channel (VIS agent) that is administered as a member of the VIS ACD split. It provides a mapping between the VIS channels and the PBX extension numbers. Refer to Figure 4-1 for a diagram of a typical application.

To open the Channel Administration screen from the ASAI Administration menu, highlight “Channel Administration” and then press **(ENTER)**. The Channel Administration screen appears. Figure 4-4 shows an example of the Channel Administration screen with channel information displayed.

Change A Channel Entry							
Channel: _____							
Extension: _____							

Please enter the new PBX extension number

HELP	CHOICES	SAVE			CANCEL		CHG-KEYS
------	---------	------	--	--	--------	--	----------

Figure 4-4. Channel Administration Screen

The Channel Administration screen contains the following fields:

- The “CHANNEL” field indicates the T/R or LST1 channel number on the VIS.
- The “EXTENSION” field indicates the Private Branch Exchange (PBX) extension number assigned for the channel.
- The “LOGIN” field indicates whether or not the channel is intended to be logged in to the ACD split. This field can be either YES or NO. If the channel is not logged in, the ACD does not deliver any calls to it.
- The “STATUS” field indicates the channel status. The status can be either broken, foos, hwoos, logout, manoos, netoos, nonex, or LOGIN. LOGIN status indicates that the T/R or LST1 channel is ready to receive calls from the ACD.

The following is a list of the various channel states. Refer to Appendix D, “Troubleshooting ASAI” for additional information on channel states:

- broken (broken) — Possible malfunction detected on line
- foos (facility out of service) — The line is not functional
- hwoos (hardware out of service) — Channel cannot be logged in because ASAI digital link is not operating
- logout (logged out) — Channel has not been administered to be logged in
- manoos (manual out of service) — Channel has not been placed into service
- netoos (network out of service) — ASAI link is up, but attempts to log in the channel are failing
- nonex (nonexistent) — Channel does not exist
- LOGIN — Channel is ready to receive calls from the ACD

The Channel Administration screen with the standard function keys is displayed in Figure 4-4. Note that you need to use **CHG-KEYS** to switch the function key display from the standard to the screen-specific set of commands, or vice versa. The screen-specific function keys in the Channel Administration screen perform the following functions:

- **ADD** — Add a channel entry. Refer to Figure 4-5 and accompanying text for additional information.
- **CHANGE** — Change the PBX extension assigned to the channel. The channel must be logged out before it may be changed. Refer to Figure 4-6 and accompanying text for additional information.
- **REMOVE** — Remove a channel entry. The channel must be logged out before it may be removed.
- **LOGIN** — Login a channel as an agent of the ACD split. You must log in the channel in order to receive calls from the ACD.
- **LOGOUT** — Logout a channel from the ACD split. You may use the **LOGOUT** key to prevent the ACD from delivering calls to the channel.

Press CANCEL to return to the ASAI Administration screen (Figure 4-3).

**NOTE:**

Once the T/R or LST1 channels are administered and logged in, no manual intervention is required to log the channels back in during recovery (for example, switch or VIS reboot) or upon restarting the voice system.

**NOTE:**

If the ASAI link goes down after a channel is logged in, the channel STATUS displays 'hwoos' even though the channel remains in a logged in state. The VIS relies on the messages sent over the ASAI link to login and logout channels. If channels are logged in, and the ASAI link abruptly goes down (for example, the cable becomes disconnected), the VIS is not able to logout the channels. (The PBX does NOT automatically logout the channels). This means that calls placed to the VIS ACD split are still directed to Tip/Ring lines on the VIS, but because the ASAI link is down, DNIS is not available for the call. In addition, the A_Callinfo and A_Tran actions do not function. To prepare for such a link outage, you should assign a 'backup' script to DNIS 'ANY' (which does not rely on A_Callinfo or A_Tran). The standard Transfer Call action may be used instead.

Add Channel Entry

The Add A Channel Entry screen may be used to assign a Tip/Ring channel as a VIS agent.

While the Channel Administration screen is active, press **ADD** to open the Add A Channel Entry screen (Figure 4-5).

Figure 4-5. Add Channel Entry Screen *achannel.ps*

The Add A Channel Entry screen contains the following fields:

- The “Channel” field indicates the Tip/Ring channel number on the VIS that you wish to add. The channel number must be unique.
- The “Extension” field indicates the extension number assigned to the application. The extension number must be unique.

Press **SAVE** to add the new channel entry or press **CANCEL** to exit the Add A Channel Entry screen and return to the Channel Administration screen.

Change Channel Entry

The Change A Channel Entry screen is used to change the extension of a VIS agent.

**NOTE:**

The channel must be logged out before it may be changed.

While the Channel Administration screen is active, press **CHANGE** to open the Change A Channel Entry screen (Figure 4-6).

Change A Channel Entry	
Channel:	_____
Extension:	_____

Please enter the new PBX extension number

HELP	CHOICES	SAVE			CANCEL		CHG-KEYS
------	---------	------	--	--	--------	--	----------

Figure 4-6. Change Channel Entry Screen

The Change A Channel Entry screen contains the following fields:

- The “Channel” field indicates the T/R or LST1 channel number on the VIS that you chose to change. The Channel field cannot be changed in this screen.
- The “Extension” field indicates the PBX extension number assigned to the channel. The extension number must be unique.

Press **SAVE** to activate the channel entry changes or press **CANCEL** to quit and return to the Channel Administration screen.

Remove Channel Entry

While the Channel Administration screen is active, press **REMOVE** to unassign a Tip/Ring channel as a VIS agent. You will receive a confirmation screen asking you if you wish to **REMOVE** the selected item.

Press **(ENTER)** to continue or **CANCEL** to abort the removal procedure.



NOTE:

The channel must be logged out before it may be removed.

Virtual Channel Administration

From the Voice System Administration screen (Figure 4-1), highlight “System Monitor”, then press **(ENTER)**. The Voice Channel Monitor screen appears (Figure 4-7)

System Monitor - Voice Channels					
Channel	Calls Today	Voice Service	Service Status	Caller Input	Dialed Digits
0	1	appl2	Talking	8509	5312
1	1	appl4	Collect		
2	3	appl7	Transfer		
3	1	appl3	DB		
4v	1	virtual1		7328	
5v	10	virtual2			

HELP	PREV-PAGE	NEXT-PAGE			CANCEL		CHG-KEYS
------	-----------	-----------	--	--	--------	--	----------

Figure 4-7. Voice Channel Monitor Screen

Routing and data screen delivery applications available with the ASAI application are data-only applications. Unlike voice response applications, such applications do not require the allocation of network or speech processing resources. For these applications, one or more “virtual channels” are allocated by the ASAI system, making all regular channels available for use by script applications that need the system resources allocated to them. At a minimum, one virtual channel is required

for each domain running a data-only script. A maximum of 32 virtual channels may be administered on the VIS.

The non-voice script that is assigned to a domain is started on the virtual channel when you use the **ENABLE** key in the Domain Administration screen to place the domain into service (inserv). Note that if the domain changes from inserv to any other state, the non-voice response script running on the virtual channel is aborted.

 NOTE:

As traffic increases on any one domain, the ASAI process spawns the assigned data-only script to accommodate the increased load.

Virtual channels are assigned a range of channel numbers above the maximum number of regular channels allocated on the system and activity on these channels is displayed in the Voice Channel Monitor screen. Following is a discussion of the fields in this screen as they relate to virtual channels used by the ASAI application. Refer to Chapter 6, “System Monitor” of the *CONVERSANT Voice Information System Operations* for additional information.

The “Channel” field lists the existing channels on the VIS. “Virtual channels” have the character ‘v’ appended to the channel number in this field. In Figure 4-7 voice.vir.ps] channels 4 and 5 are virtual channels.

The “Calls Today” field provides data on the number of calls made to a particular channel within the system. Calls are monitored for a 24-hour period, on any day beginning at midnight (12 a.m.). For virtual channels, the value in this field is the number of times the channel was used to run a data-only script. In Figure 4-7 channels 4 and 5 are running the virtual1 and virtual2 data-only scripts, respectively.

The “Voice Service” field provides the name of the service currently running on the corresponding voice channel. In Figure 4-7 channel 4 is running the virtual1 service and channel 5 is running the virtual2 service. For virtual channels, this field is blank when no script is running.

The “Service Status” field lists the various status states of each channel in the VIS. The Service Status field is blank for virtual channels except when a script with a Data Interface Process (DIP) is running, in which case this field indicates the name of the DIP.

The “Caller Input” field provides the last set of digits entered by the caller. The Caller Input field is always blank for virtual channels.

The “Dialed Digits” field provides the last set of digits dialed by the VIS during the transfer process. The Dialed Digits field is always blank for virtual channels.

Diagnose IPCI Board

You should use the Diagnose IPCI Board command when you suspect that there is a problem with the ASAI link. Note that the IPCI card must be on-line before it may be diagnosed. Refer to the Initialize IPCI Board procedures later in this chapter for information on initializing the IPCI card.

You may use the Show Status of ASAI Link (Figure 4-15) screen to determine whether the link is on-line (actively using the link) or off-line (not using the link). You may use the Take IPCI Board Off-line procedure to change the card from an on-line to an off-line state. Likewise, you may use the Initialize IPCI Board procedure to change the card from an off-line to an on-line state. The Take IPCI Board Off-line and Initialize IPCI Board procedures are discussed later in this chapter.

To diagnose the IPCI Board from the ASAI Administration screen, highlight “Diagnose IPCI Board” and then press **ENTER**. Figure 4-8 shows the Diagnose IPCI Board screen with IPCI diagnosis information displayed.

Diagnose IPCI Board			
The IPCI Board passed all tests.			
Press ENTER to continue.			

	PREVPAGE	NEXTPAGE		CANCEL	CHG-KEYS
--	----------	----------	--	--------	----------

Figure 4-8. Diagnose IPCI Board Screen

In the Diagnose IPCI Board example shown in Figure 4-8, testing was executed on card 1. If for some reason diagnostics had failed on the card, that information would be contained in this screen. The diagnose IPCI card command does not affect the link or traffic on the link.

Domain Administration

The Domain Administration screen is used to instruct the VIS adjunct on how to handle a call that was offered to a specific domain. For voice-response applications, there may be only one entry of type ACD to which the VIS service is assigned.

Other domain types that may be administered through the Domain Administration screen and assigned only non-voice response (data only) services include the following:

- ACD — A domain type which processes information from the PBX associated with calls distributed on an ACD split on the switch.
- VDN — A domain type which processes information from the PBX associated with a call processing vector (VDN) on the switch.
- RTE — A domain type which processes route requests from the switch.
- CTL — A domain type which processes information from the PBX for calls transferred from the VIS (using the A_Tran action) to an extension which is not associated with an ACD or VDN domain already monitored by the VIS.

To open the Domain Administration screen from the ASAI Administration screen, highlight “Domain Administration” and then press **(ENTER)**. The Domain Administration screen appears. Figure 4-9 shows an example of the Domain Administration screen with domain information displayed.

Domain Administration				
NAME	TYPE	EXT	SERVICE	STATUS
ACD001	ACD	3982	VIS	inserv
VDN001	VDN	4002	vdn-serv	manoos
CTL001	CTL	ANY	ctl-serv	inserv
RTE001	RTE	4005	rte-serv	inserv
ACD002	ACD	7392	acd-serv	inserv

Please highlight the item you want and press a function key.

HELP					CANCEL		CHG-KEYS
------	--	--	--	--	--------	--	----------

Figure 4-9. Domain Administration Screen **domain.ps**

By default, the VIS updates the Domain Administration screen every 2 seconds. The Domain Administration screen contains the following fields:

- The “NAME” field indicates the domain name. You may choose any name for the domain name.
- The “TYPE” field indicates the VIS domain type. The domain TYPE can be one of the following:
 - ACD — Domain on the VIS which monitors calls to the corresponding split (domain) on the PBX.
 - VDN — Domain on the VIS which monitors calls to the corresponding VDN (domain) on the PBX.
 - CTL — Defines a domain on the VIS which monitors calls transferred away from the VIS (by a voice script using the A_Tran action) to destinations on the PBX that are not monitored by an ACD or VDN domain (for example, monitor calls transferred using A_Tran to miscellaneous extensions). CTL domains are defined by only the VIS and do not correspond to any domain on the PBX.
 - RTE — Defines a domain on the VIS which accepts Route Requests from the PBX. RTE domains are defined by only the VIS and do not correspond to any domain on the PBX.

The meaning of the “EXT” field depends on the type of domain. The EXT can be one of the following:

- ACD type domains — EXT field should contain the corresponding ACD split PBX extension that is being monitored.
- VDN type domains — EXT field should contain the corresponding VDN PBX extension that is being monitored.
- CTL type domains — EXT field designates an extension (which does not correspond to an ACD or VDN extension) for which calls are being transferred by a VIS channel using the A_Tran action. Calls transferred to the specified extension are processed by the CTL domain. The extension field can also contain “ANY,” in which case, calls transferred to any destination not already monitored by another domain are processed. Note that if a specific extension is used in this field, it must correspond to an extension used in the Destination field of the A_Tran action used by a script assigned to an ASAI channel.
- RTE type domains — EXT field limits the processing of route requests based on the extension that was dialed. This field can contain a specific extension or the word “ANY.” If an extension is specified, only route requests for the specified (called) extension are processed. If “ANY” is specified in the field, then all route requests not processed by any other administered domain are processed.

The "SERVICE" field indicates how the calls offered to the domain are handled by the VIS. The SERVICE field is used to specify a script which services the domain.

⇒ NOTE:

To make sure that the latest version of a script is picked up, the corresponding domain should be disabled and enabled each time the script is verified and installed via Script Builder.

A script can be assigned to any type of domain (ACD, VDN, etc.). If the application or the ACD directs calls to the VIS T/R or LST1 line, the special service "VIS" must be used here. The SERVICE can be one of the following:

- ACD type domains — The SERVICE can be either a monitoring script (refer to Chapter 2, "ASAI Application Planning and Design", for a description of a monitoring script) or the special word "VIS." If the agents on the ACD split are Tip/Ring channels on the VIS (that is, channels administered in the Channel Administration screen described in Chapter 2, "ASAI Application Planning and Design"), then you should specify "VIS" as the service for the domain. "VIS" service provides the ability to start voice scripts on the Tip/Ring channels based on the DNIS. It also provides the ability for those voice scripts to use the A_Callinfo action.

⇒ NOTE:

VIS can be assigned to only one ACD domain. Consequently, all Tip/Ring channels that are administered for ASAI must be an agent that belongs to this ACD domain.

For ACD type domains that do not have agents on the VIS, a monitoring script should be assigned in this field.

- VDN and CTL type domains — The SERVICE must be a monitoring script. (refer to Chapter 2, "ASAI Application Planning and Design" for a description of a monitoring script).
- RTE type domains — The SERVICE must be a routing script. (Refer to Chapter 2, "ASAI Application Planning and Design" for a description of a routing script).

The VIS T/R or LST1 channels (agents) are used to support voice response applications. All T/R or LST1 channels used for ASAI capability must be administered into one ACD split on the PBX. Calls offered to the VIS domain are handled by the Script Builder script associated with the DNIS information. Refer to Chapter 3, "Configuration Management" in *CONVERSANT Voice Information System Operations* for information on how to administer scripts to dialed numbers.

The “STATUS” field indicates whether the domain is ready to receive call information. The domain STATUS can be one of the following. Refer to Appendix D, “Troubleshooting ASAI” for additional information.

- broken (broken) — A virtual channel could not be allocated for the service assigned to this domain.
- foos (facility out of service) — The ASAI digital link is not operating.
- initing (initializing) — The service assigned to the domain is failing initialization.
- inserv (in service) — The domain is ready to receive call information from the switch.
- manoos (manual out of service) — The domain has not been placed into service.
- netoos (network out of service) — The ASAI link is up, but attempts to receive call information from the switch are failing.

The Domain Administration screen with the standard function keys is displayed in Figure 4-9. Note that you need to use **CHG-KEYS** to switch the function key display from the standard to the screen-specific set of commands, or vice versa. The screen-specific function keys in the Domain Administration screen perform the following functions:

- **ADD** — Add a domain entry. Refer to Figure 4-10 and accompanying text for additional information.
- **CHANGE** — Change a domain entry. The domain must be disabled before it can be changed. Refer to Figure 4-11 and accompanying text for additional information.
- **REMOVE** — Remove a domain entry. The domain must be disabled before it can be removed.
- **ENABLE** — Place domain into service. When enabled, the domain is monitored and the VIS receives events for the domain. If the enable request is successful, the domain status becomes inserv. The domain must be disabled before it can be enabled.
- **DISABLE** — Take domain out of service. If the disable request is successful, the domain status becomes manoos.

Press **CANCEL** to return to the ASAI Administration screen (Figure 4-3). Once the T/R or LST1 channels are administered and logged in, no manual intervention is required to log the channels back in during recovery (for example, switch or VIS reboot) or on restarting the voice system.

Once the domain is administered and made inserv, no manual intervention is required to bring the domain back into service during recovery (for example, switch or VIS reboot) or upon restarting the voice system.

If a VIS domain with a script assigned to it is in any state but inserv, the default script is invoked. Note that if the domain for the VIS agent (T/R or LST1) line is disabled, the VIS still takes calls on these lines.

Add Domain Entry

While the Domain Administration screen is active, press ADD to open the Add A Domain Entry screen (Figure 4-10).

Add A Domain Entry	
Name:	_____
Type:	_____
Ext :	_____
Service:	_____

Please enter the domain name

HELP	CHOICES	SAVE			CANCEL		CHG-KEYS
------	---------	------	--	--	--------	--	----------

Figure 4-10. Add Domain Entry Screen

The Add A Domain Entry screen contains the following fields. Refer to Figure 4-9 and accompanying discussion for more information on each of these fields.

- The “Name” field indicates the name of the domain you wish to add.
- The “Type” field indicates the domain type.
- The “Ext” field indicates the extension number assigned to the application.
- The “Service” field indicates how the calls offered to the domain are handled by the VIS.

Press **SAVE** to add the new domain entry or press **CANCEL** to exit the Add A Domain Entry screen and return to the Domain Administration screen.

Change Domain Entry

While the Domain Administration screen is active, press **CHANGE** to open the Change A Domain Entry screen (Figure 4-11).

**NOTE:**

The domain must be disabled before it can be changed.

Change A Domain Entry	
Name:	_____
Type:	_____
Ext :	_____
Service:	_____

Please enter the domain type: ACD, VDN, CTL, or RTE.

HELP	CHOICES	SAVE			CANCEL		CHG-KEYS
------	---------	------	--	--	--------	--	----------

Figure 4-11. Change Domain Entry Screen

The Change A Domain Entry screen contains the following fields:

- The “Name” field indicates the name of the domain you wish to change.
- The “Type” field indicates the domain type.
- The “Ext” field indicates the extension number assigned to the ACD split which uniquely identifies the domain in the switch.
- The “Service” field indicates how the calls offered to the domain are handled by the VIS.

Press **SAVE** to activate the domain changes or press **CANCEL** to quit and return to the Domain Administration screen.

Remove Domain Entry

While the Domain Administration screen is active, press **REMOVE**. You will receive a confirmation screen asking you if you wish to **REMOVE** the selected item. Press **(ENTER)** to continue or **CANCEL** to abort the removal procedure.



NOTE:

The domain must be disabled before it can be removed.

Initialize IPCI Board

The IPCI Board must be initialized when it is taken off-line. The Initialize IPCI Board procedure downloads the driver software onto the card and puts the card on-line. The IPCI card is now actively using the link.



WARNING:

The "Initialize IPCI Board" command should be used only as an error recovery procedure for the card. You do not need to initialize the IPCI card each time that you diagnose the card.



NOTE:

If the IPCI card is already initialized and on-line, the card must first be taken off-line before it can be re-initialized.

The Initialize IPCI Board screen contains information on what occurred during initialization of the IPCI card. The Initialize IPCI Board screen displays information as to whether the initialization was a success or failure and the reason for the failure. Figure 4-12 displays an example of the Initialize IPCI Board screen with IPCI card initialization information displayed.

To initialize the IPCI card from the ASAI Administration screen, highlight "Initialize IPCI Board", then press **ENTER**.

Initialize IPCI Board			
IPCI Board Initialization completed successfully.			
Press ENTER to continue.			

PREVPAGE	NEXTPAGE	CANCEL	CHG-KEYS
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Figure 4-12. Initialize IPCI Board Screen

Parameter Administration

The ASAI Parameters screen is provided to allow adjustment of ASAI system parameters. These parameters affect the behavior of the ASAI feature.

To open the ASAI Parameters screen from the ASAI Administration screen, highlight “Parameter Administration” and then press **ENTER**. The ASAI Parameters screen appears (Figure 4-13).

ASAI Parameters
CONNECT Event: CONNECTED
Trace Detail: Low

Enter when the CONNECT Event should be reported to A_Event

HELP	CHOICES	SAVE			CANCEL		CHG-KEYS
------	---------	------	--	--	--------	--	----------

Figure 4-13. ASAI Parameters Screen *param.ps*

The ASAI Parameters screen contains the following fields:

The “CONNECT Event” field is used to specify when the CONNECT event is reported to the A_Event action in a script assigned to an ACD, VDN, or CTL type domain. Valid entries for this field are ‘CONNECTED’ or ‘ALERTING’. The default for this field is CONNECTED.

If you select **CONNECTED** in the **CONNECT** Event field, the **CONNECT** event is reported when the ASAI **CONNECTED** message is received from the PBX. Typically, this corresponds to when the call is answered. If you select **CONNECTED**, **VIS** Data is transferred to subsequent **CONNECT** event reports when a call is transferred from one live agent to another (provided the transfer is a blind transfer). Refer to the discussion of the **VIS** Data field in the **A_Trans** and **A_Event** actions in Chapter 6, "ASAI Script Builder Actions" for additional information. Note that **VIS** Data is always transferred when the **A_Trans** action transfers the call from a Tip/Ring line on the **VIS** to another number, regardless of whether **CONNECTED** or **ALERTING** is chosen for this field.

If you select **ALERTING** in the **CONNECT** Event field, the **CONNECT** event is reported when the call starts to ring at the destination number. This corresponds to the receipt of the ASAI **ALERTING** message from the PBX. In cases where a **CONNECTED** message is received from the PBX (the call is answered) and an **ALERTING** message was not received previously, the **CONNECT** event is reported to **A_Event** upon receipt of the **CONNECTED** message. This can happen when the call is routed to an ISDN trunk. An advantage of choosing **ALERTING** is that if your application performs data screen delivery to agents, the screen can be displayed before the agent answers the call. This gives the host computer and agent more time to process the data before conversing with the customer.

⇒ NOTE:

Typically, neither a **CONNECTED** nor **ALERTING** message is received from the PBX when calls are placed to non-ISDN trunks, thus a **CONNECT** event is not reported to **A_Event** for the call. However, an **END** event is reported when the call ends.

⇒ NOTE:

The **CONNECT** event parameter is a system parameter and, thus, affects all scripts using the **A_Event** action.

The "Trace Level" field controls the amount of detail that is displayed when you use the trace **dip7** command to monitor messages and events being processed by the ASAI system. The trace feature facilitates the debugging of new applications and is an optional feature that is not required for normal system operation.

Valid entries for this field are 'Low', 'Normal', and 'High'. The Trace Level default setting is Low. This default setting displays only ASAI error and warning conditions and should be used when there is live traffic to minimize processing overhead from the trace feature. Refer to Table 4-1 for information on Trace detail display.

The Normal setting can be used for simple debugging of application scripts which use the A_Callinfo, A_Event, A_RouteSel, and A_Tran actions. Normal detail causes trace to display Low detail information as well as information pertaining to the processing of the A_Callinfo, A_Event, and A_RouteSel, and A_Tran script actions. This information may be useful when debugging a new application script. The format is specific to each ASAI action being processed.

The High setting gives additional information on ASAI messages that are sent and received between the VIS and the DEFINITY Generic 3i system. High detail causes trace to display Low and Normal detail as well as ASAI messages (call events) and routing messages. ASAI messages received from the PBX which contain information about a call on a domain. This information may be useful when debugging an application script which is monitoring the progress of calls on the PBX. S to Appendix D, "Troubleshooting ASAI" for additional information.

Table 4-1. Trace Detail Display

Trace Detail	Trace Display
Low	— ASAI error and warning condition
Normal	— Information displayed by Low setting — ASAI script action (that is, A_Callinfo, A_Tran, A_Event, and A_RouteSel) processing information
High	— Information displayed by Low and Normal Settings — ASAI messages (call events) received from the PBX which contains information about a call on a domain

Show ASAI Software Version

The Show ASAI Software Version screen is a text screen which contains information on the software versions loaded onto the VIS. To show the ASAI Software Version from the ASAI Administration screen, highlight “Show ASAI Software Version”, then press **ENTER**. Figure 4-14 displays an example of the Show ASAI Software Version screen with software version information displayed.

Figure 4-14. Show ASAI Software Version Screen *version.ps*

Show Status ASAI Link

The Show Status of ASAI Link screen provides information on the current status of the ASAI link to the VIS. To show the status of the ASAI link from the ASAI Administration screen, highlight “Show Status of ASAI Link”, then press **ENTER** to display link status information. Figure 4-15 displays the Show Status of ASAI Link screen for an IPCI card and ASAI link which are functioning normally.

Show Status of ASAI Link			
The IPCI Board is on-line. Physical Layer (L1) is UP, Link Layer (L2) is UP.			
Press ENTER to continue.			

	PREVPAGE	NEXTPAGE			CANCEL		CHG-KEYS
--	----------	----------	--	--	--------	--	----------

Figure 4-15. Show Status ASAI Link Screen

The Show Status of ASAI Link screen displays one of the messages listed in Table 4-2 if the link is experiencing problems. For each of the displays in the Show Status of ASAI Link screen listed below, a meaning is provided which corresponds to the first column of Table D-3. Refer to Appendix D, “Troubleshooting ASAI” for possible remedies for each problem.

Table 4-2. Show Status ASAI Link Displays

Show Status of ASAI Link	Meaning
The IPCI Board is on-line. Physical layer (L1) is DOWN, Link Layer (L2) is DOWN	L1 DOWN, L2 DOWN
The IPCI Board is on-line. Physical layer (L1) is UP, Link Layer (L2) is DOWN	L1 UP, L2 DOW
The IPCI Board may be faulty or non-existent. Make sure the card and the ASAI Library package have been installed with the correct options.	FAULTY HARDWARE or UNKNOWN or UNDETERMINED
The IPCI Board is currently OFFLINE. The Board must be initialized before service can begin.	OFFLINE
The IPCI Board is currently being initialized.	PUMPING
The IPCI Board is currently being taken OFFLINE	AWAITING GOING OFFLINE

Take IPCI Board Off-line

The Take IPCI Board Off-Line procedure takes the IPCI card off-line and effectively disables the link between the IPCI card and the driver software. If you suspect that there might be problems with the ASAI communications and you would like to reinitialize the IPCI card, you must take the card off-line by executing the following command before you may execute the Initialize the IPCI card command.

**WARNING:**

Do not take the card off-line while the ASAI application is running. This command should not be used in normal operation situations. It should be used in conjunction with the "Initialize IPCI Board" command as an error recovery procedure for the card.

From the ASAI Administration screen, highlight "Take IPCI Board Off-line", then press **ENTER** to execute the IPCI card off-line procedure. Figure 4-16 shows an example of the Take IPCI Board Off-Line screen when the card is successfully taken off-line.

Take IPCI Board Off-line						
IPCI Board has been taken off-Line.						
Press ENTER to continue.						

	PREVPAGE	NEXTPAGE			CANCEL		CHG-KEYS
--	----------	----------	--	--	--------	--	----------

Figure 4-16. Take IPCI Board Off-line Screen

The Take IPCI Board Off-line screen contains information on whether the IPCI card has been successfully taken off-line.

ASAI Administration Overview

Administration of the ASAI feature is a three-step procedure performed through the VIS menu system. The following administration procedures assume you are installing a voice response application with a configuration in which calls placed to an Automatic Call Distributor (ACD) on the Private Branch Exchange (PBX) are directed to (agent) lines on the VIS. The VIS is to select a service for the incoming call based on the dialed number (DNIS). The service requests the dialed number and calling number (ANI) from the ASAI and uses this information as part of the service being provided to the caller. The steps to be performed on the switch and the adjunct are as follows:

5. Administer the T/R or LST1 telephone lines.
6. Administer the VIS ACD domain
7. Administer the lines as agents on the ACD split and log in the lines.

Once these steps have been completed, you can assign services to called numbers. Refer to Chapter 3, "Configuration Management" of the *CONVERSANT Voice Information System Operations* for information on performing this procedure.

NOTE:

The following assumes that you have completed the necessary administration on the PBX. Please refer to the *DEFINITY Communications System Generic 3i Implementation*, 555-230-650 for additional information.

Administering the Lines

The lines (T/R or LST1) are administered as described in Chapter 5, “Switch Interface Administration” of *CONVERSANT VIS Version 4.0 Operations*, 585-350-703. To be certain that you select options that are compatible with the DEFINITY Generic 3i (only certain versions) system, select “AT&T System 75” item in the PBX Defaults screen. AT&T System 75 is the default setting. Consequently, if you are administering a new system, the lines are configured correctly by default.

Place all the lines into service. To do so, refer to the information on changing maintenance state in the operations book, Chapter 3, “Configuration Management”. Do not proceed until the lines have been placed in the inserv state. These lines or channels are referred to in the following text as *VIS Agent lines*.

Administering the VIS ACD Split Domain

In order for the VIS to receive information on calls placed to its agent lines, the ASAI feature must be administered to monitor the ACD hunt group Extension. In other words, you must administer the ASAI feature on the VIS so that it requests call events (information) from a *domain* on the PBX. In this case, the domain is the ACD hunt group or split, which is composed of the VIS agent lines. This domain is referred to as the *VIS ACD domain*. You may administer only one VIS ACD split domain on the system. Therefore, all VIS agent lines must be part of a single ACD split. Refer to Figure 3-1 for a diagram showing this configuration. The following steps outline the administration of this Domain.

1. From the Voice System Administration menu (Figure 4-1), highlight “Feature Packages” and then press **(ENTER)**. A menu of installed feature packages should be displayed in the Feature Packages screen as shown in Figure 4-2.
2. From the Feature Packages menu, highlight “ASAI Administration” and then press **(ENTER)** to display the ASAI Administration screen (Figure 4-3).
3. From the ASAI Administration menu, highlight “Domain Administration” and then press **(ENTER)** to display the Domain Administration screen. This screen is similar to that shown in Figure 4-9 except that there no domain entries.
4. Press **CHG-KEYS** to display the alternate set of function keys and then press **ADD** to create a new domain. The Add A Domain Entry form is displayed (Figure 4-10).

5. Select any name you choose and enter it into the "Name" field. Although it is not necessary, you may want to choose the same name given to the split on the PBX. To display the PBX name for the split, type **list hunt** at the DEFINITY Generic 3i system console. Find the hunt group with the group extension corresponding to your VIS ACD extension. Use the name (if any) shown for that group.
6. Enter ACD in the "Type" field.
7. Enter the group extension of the VIS ACD split in the "Ext" field.
8. Enter VIS in the "Service" field. By choosing VIS as the service, you have designated this ACD split domain as the VIS ACD split domain. This instructs the ASAI to send call information it receives from this domain to the voice response service(s) that service the VIS Agent lines.
9. Press **SAVE** to save the domain and its parameters. The Domain Administration screen should be redisplayed, similar to that shown in Figure 4-9, but now showing the domain that you have just entered.
10. Press the cursor keys until the domain is highlighted. Press **ENABLE** to bring the domain into service. Once you have pressed **ENABLE**, the VIS automatically enables the domain should the voice system be restarted after a system shutdown or power loss. You do not need to use the **ENABLE** key again to enable the domain unless you have disabled the domain by pressing the **DISABLE** key. The STATUS field should change to inserv.
11. If the STATUS field shows inserv, proceed to the information Administering the VIS Agent Lines in this chapter. If the STATUS field does not show inserv, refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting a domain.

Administering the VIS Agent Lines

After creating and bringing the VIS ACD split domain into service, you must next administer and log in the lines as VIS agent lines. This is required if your service is going to use DNIS or the A_Callinfo or the A_Tran actions described in Chapter 6, "ASAI Script Builder Actions". If an agent line is not logged in, the PBX ACD does not route any calls to it. (Note that you can still dial the agent line directly, but no call information is available to the service that answers the call. In other words, the A_Callinfo action does not return any information for a call that is not routed to the VIS by the ACD.) The following steps outline the procedure to administer and log in lines as VIS agent lines.

1. From the ASAI Administration menu (Figure 4-3), highlight "Channel Administration" and then press **(ENTER)** to display the Channel Administration screen. The Channel Administration screen should be displayed as shown in Figure 4-4, but without any entries.
2. Press **CHG-KEYS** to display another set of function keys. Press **ADD** to display the Add a Channel Entry form (Figure 4-5).
3. Enter the channel number of one of the agent lines in the "Channel" field.
4. Enter the corresponding PBX extension number of the agent line in the "Extension" field.
5. Press **SAVE** to save the parameters. The Channel Administration screen should be redisplayed with the parameters you have just entered. The LOGIN field should show NO and the STATUS field should show logout.
6. Repeat the previous steps for each of the VIS Agent lines.
7. In the Channel Administration screen (Figure 4-4), press **CHG-KEYS** to display the other set of function keys. Press the cursor keys to highlight the first channel. Press **LOGIN** to log in the VIS agent line. Once you have pressed the LOGIN key, the VIS remembers this and automatically logs in the channel should the voice system be restarted after a system shutdown or power loss. You do not need to use the LOGIN key again to log in the channel unless you have logged out the channel by pressing the **LOGOUT** key. The LOGIN field should change from NO to YES, and the STATUS field should change from logout to LOGIN.
8. If the LOGIN field shows YES and the STATUS field shows LOGIN, the channel is ready to receive calls from the ACD. If the LOGIN field still shows NO, try pressing the **LOGIN** key again. If the LOGIN field shows YES but the STATUS field does not show LOGIN, refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting a VIS agent line.
9. Repeat the previous procedures to log in the remaining channels.

Once all the channels have been logged in, you are ready to run applications. If you have not done so already, you may now assign DNIS service to channels using the Assign Service To Voice Channels screen and Assign Services to Called Numbers screen as described in *CONVERSANT VIS Version 4.0 Operations*, 585-350-703, Chapter 3, "Configuration Management."

For more information on defining a transaction and other Script Builder actions, refer to Chapter 4, “Defining the Transaction” in the *CONVERSANT VIS Script Builder*, 585-350-704. For additional information on actions, refer to Chapter 10, “Using Advanced Features” of the Script Builder book.

ASAI Script Builder Actions Overview

This chapter contains information on the actions used to access the AT&T Adjunct/Switch Application Interface (ASAI) capabilities from the Script Builder environment.

The following actions are listed in alphabetical order:

- **A_Callinfo** — Used to access call information obtained from ASAI for a call on a VIS Agent line.
- **A_Event** — Retrieves information related to a call being monitored by an ASAI domain.
- **A_RouteSel** — Used to send an ASAI route select message to the Private Branch Exchange (PBX).
- **A_Tran** — Used by voice scripts running on VIS Agent (T/R or LST1) lines to transfer a call to a live ACD, live agent, or other station on the PBX.

Defining A_Callinfo

The A_Callinfo action is used to access call information obtained from ASAI for a call on a VIS Agent line. The A_Callinfo action returns the Calling Party Number and Called Party Number associated with an incoming call to the transaction script environment. The A_Callinfo action also returns a Call ID which identifies the call. In addition, if touch-tone digits entered by the caller have been collected by the switch, A_Callinfo returns these digits. If the call was delivered to a switch through a trunk and Automatic Number Identification (ANI) information for the call is not available, the Trunk Group Id is returned instead of the Calling Party Number.

To add A_Callinfo to a transaction while defining an application in Script Builder, press **ADD**. The Action Choices screen opens. In the Action Choices screen, highlight "A_Callinfo", then press **ENTER**. Press **CANCEL** to exit from the Action Choices screen. You must now define the A_Callinfo step further. In the Define Transaction screen, highlight "External Action: A_Callinfo," then press **DEFINE**. The Define A_Callinfo screen opens.

Figure 6-1. Define A_Callinfo Form Screen def_acall.ps

Each of the fields in the Define A_Callinfo screen must contain a field name which returns the following information. Refer to Table 6-1 for a summary of the information in each of these fields.



CAUTION:

You should specify only 'num' field types for those fields in the Define A_Callinfo screen that return numbers, that is, specify 'num' fields or constants for those fields that are of type 'num.'

The "Calling Party Number" field stores the calling party number. The value returned in the Calling Party Number field can be up to 20 characters in length. If the A_Callinfo action is successful, this field contains the calling party number. If the calling party number is not known or the call was routed from a non-ISDN trunk, a string of length 0 (null value) is returned.

The "Called Party Number" field stores the called party number. The value returned in the Called Party Number field can be up to 20 characters in length. If the Called Party Number is not known, a string of length 0 (null value) is returned.

The "Switch Data" field stores up to 16 characters. If the switch prompts the caller for touch-tone digits, the digits are returned in the Switch Data field. If no digits are collected by the switch, a string of length 0 (null value) is returned.

The "Trunk Group Id" field indicates whether the incoming call was routed from a non-ISDN trunk on the switch. If the call was not placed through a trunk, the value returned is 0.

The "Call Id" field stores the Call Id (assigned by the switch) that identifies the incoming call. If the Call Id is not known, the value returned is 0.

The "Cause Value" field returns an error cause if the request for call information is not successful. Note that the error cause is returned if the Return Field contains a value of -3. Refer to VIS value column in Table D-6 in Appendix D, "Troubleshooting ASAI" for a complete listing of cause values.

The "Return Field" holds the return status of the A_Callinfo action. If the A_Callinfo action is successful, it returns a number greater than or equal to zero. If A_Callinfo is unsuccessful, it returns one of the following negative values:

- -3 — ASAI could not process the request for information. Check the Cause Value field for information on why the request failed.
- -1 — A_Callinfo could not send the request for call information. Check the Message Log Report for system errors. Refer to Chapter 4, "Reports Administration" in the *CONVERSANT Voice Information System Operations* for additional information.
- -2 — A_Callinfo did not receive a response from the ASAI for the request for information. Check to see if the ASAI system is running.
- -4 — ASAI link is down and call information cannot be received from the switch. Refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting the ASAI digital link.
- -5 — Illegal request. The call is not being monitored so no call information is available. Make certain that A_Callinfo is only being used in a script assigned to a VIS agent line and that the domain with which the call is associated is being monitored by the VIS. Refer to Chapter 4, "ASAI Administration" for information on Channel Administration and Domain Administration, including how to monitor a channel.
- -6 — Switch did not respond with the call information.

After the user-defined entries for the Define A_Callinfo screen are completed, press **CLOSE** to update the "External Action: A_Callinfo" action step in the Define Transaction list.

Table 6-1. A_Callinfo Fields

Field	Input/Output	Required?	Field Type	Field Size
Calling Party Number	output	required	char	20
Called Party Number	output	required	char	20
Switch Data	output	required	char	16
Trunk Group ID	output	required	num	–
Call Id	output	required	num	–
Cause Value	output	required	num	–
Return Value	output	optional	num	–

Defining A_Event

The A_Event action retrieves information related to a call being monitored by an ASAI domain. This action must be used with all scripts assigned to a domain. (Refer to Chapter 4, "ASAI Administration" for information on administering a domain). The A_Event action can return the following types of events:

- Abandon Events — a monitored call is dropped or caller hung up
- Connect Events — a monitored call is being delivered to an agent
- End Events — a monitored call has ended normally (that is, not abandoned)
- Route Request Events — a switch has requested routing for a particular call
- Abnormal Route End Events — the switch is reporting that the routing failed due to some reason specified in the Cause Value field.

**CAUTION:**

The A_Event action should not be used in a script assigned to a voice channel.

To add A_Event to a transaction while defining an application in Script Builder, press **ADD**. The Action Choices screen opens. Highlight "A_Event," then press **ENTER**. Press **CANCEL** to exit from the Action Choices screen.

You must now define the A_Event step further. In the Define Transaction screen, highlight “External Action: A_Event,” then press **DEFINE**. The Define A_Event screen opens.

Define A_Event							
Connected Party Number: _____							
Calling Party Number: _____							
Called Party Number: _____							
Switch Data: _____							
Trunk Group Id: _____							
Call Id: _____							
Other Call Id: _____							
LAI Display Info: _____							
VIS Data: _____							
Cause Value: _____							
Cause Value: _____							
Return Field: _____							

Enter a field name to store the connected party number.

HELP	CHOICES	CLOSE	REDRAW		CANCEL		
------	---------	-------	--------	--	--------	--	--

Figure 6-2. Define A_Event From Screen

Each of the fields in the Define A_Event screen must contain a field name which returns the following information. Refer to Table 6-4 for a summary of the information in each of these fields.



CAUTION:

You should specify only 'num' field types for those fields in the Define A_Event screen that return numbers, that is, specify 'num' fields or constants for those fields that are of type 'num.'

The “Connected Party Number” field returns a number or extension, depending on the event being reported. For the ‘C’ (Connect) event, the field returns the alerted extension (answering extension) if the ‘C’ event is triggered on the alerting or connect message from the switch. If A_Event indicates the call has ended (the Return Field returns a value of ‘E’), the Connected Party Number field contains the number of the last connected or alerted party. If A_Event indicates the call is abandoned (the Return Field returns a value of ‘A’), the Connected Party Number field returns an alerted agent extension or a string of length 0 (null value) depending on when the call was abandoned. If the call was abandoned before agent selection, the Connected Party Number field returns a string of length 0 (null value). If the call was abandoned after agent selection, the Connected Party Number field returns the extension of the agent that was alerted for the call prior to the abandon. If the connected party number is not known, a string of length 0

(null string) is returned. If A_Event is reporting a route request (the Return Field returns a value of 'R'), a string of length 0 (null value) is returned. The value returned can be up to 20 characters in length.

The "Calling Party Number" field stores up to 20 characters. If the A_Event action is successful, this field contains the calling party number. If the calling party number is not known or the call was routed from a non-ISDN trunk, a string of length 0 (null value) is returned. If the value returned from the Other Call Id field is not 0, then the Calling Party Number returns the number of the original calling party.

The "Called Party Number" field stores the called party number. The value returned in the Called Party Number field can be up to 20 characters in length. If the Called Party Number is not known, a string of length 0 (null value) is returned. If the value returned from the Other Call Id field is not 0, then the Called Party Number returns the number that was originally dialed.

The "Switch Data" field stores up to 16 characters. If the switch prompts the caller for touch-tone digits, the digits are returned in the Switch Data field. If no digits are collected by the switch, a string of length 0 (null value) is returned.

The "Trunk Group Id" field indicates whether the incoming call was routed from a non-ISDN trunk on the switch. If the call was not placed through a trunk the value returned is 0.

The "Call Id" field stores the Call Id (assigned by the switch) that identifies the incoming call. If the Call Id is not known, the value returned is 0. The "Other Call Id" field returns a number (assigned by the switch) that identifies the original call that was transferred. The script developer may use this field to associate subsequent events that occur from a previous call. If the field is 0, then A_Event cannot relate this event with any other call. If the call is a new incoming call to a monitored domain, then the field is always 0. If it is a transfer call, the field value depends on the type of transfer that was completed. A consult transfer returns a non-zero value only when the transfer call is directed to a non-monitored domain. A blind transfer always returns a non-zero value. Refer to the information on agent-to-agent transfers in Chapter 2, "ASAI Application Planning and Design" for additional information.

The "LAI Display Info" field returns the Look Ahead Interflow (LAI) display information for the call. The value returned in this field depends on the call flow and whether or not the LAI feature (a PBX feature) is used. When the LAI feature is used to connect calls from one call center to another, this information can be used to determine which call center application was handling the caller on the originating switch. The originating switch can be administered such that the LAI Display Info field will contain the originally dialed number the caller used at the originating switch. If the LAI feature is not used to connect calls from one switch to another, then a string of length 0 (null value) is returned.

The “VIS Data” field returns a value previously saved in the VIS Data field of the A_Tran action in a voice script. If the call was not previously transferred using A_Tran, then a string of length 0 (null value) is returned. For example, a voice script which responds to an incoming call to the VIS (on a channel administered for ASAI) collects information from the caller and saves information in the VIS Data field of the A_Tran action and then uses A_Tran to transfer the call to a domain administered on the VIS. When A_Event reports the Abandon, Connect, or End events (Return Field contains 'A', 'C', or 'E') for that call, this field returns the saved information.

The “Routing ID” field contains a unique number that identifies the route request if A_Event is indicating a route request (that is, the Return Field contains an 'R'). This number is needed if the A_RouteSel action is used to respond to the route request. If the Routing ID is not known, the value returned is 0.

The “Cause Value” field returns an error cause from the switch if the request to route the call is not successful. Note that the error cause is returned if the Return Field contains a value of “r” or 114 (indicating an ABNORMAL ROUTE END event). Possible error causes include the following:

- “0” — The Destination Number provided in the A_RouteSel action is invalid and does not exist on the switch.
- “1” — The switch is unable to route the call to the Destination Number because the destination is busy (that is, the destination currently has an active call).
- “8” — The call dropped while waiting for a routing response. The caller probably hung up before the call has been routed.
- “12” — The vector processing on the switch encountered steps other than *wait*, *announcement*, *goto*, or *stop* after the adjunct routing command.
- “13” — Upon routing to the Destination Number (for direct agent call), the Destination Number is not logged in to the specified Split Extension.
- “14” — The Destination Number (for direct agent call) is not a member of the specified Split Extension in the A_RouteSel action.

The “Return Field” contains a return code indicating what type of event (if any) is being reported. If the A_Event action is successful, it returns a number greater than or equal to zero. Refer to Table 6-4 for additional information. If A_Event is unsuccessful, it returns one of the following values.

- -1 — A_Event could not send the request for call information. Check the Message Log Report for system errors. Refer to Chapter 4, “Reports Administration” in the *CONVERSANT Voice Information System Operations* for additional information.
- -2 — A_Event did not receive a response from the ASAI for the request for information. Check to see if the ASAI system is running.

- -4 — ASAI link is down and call information cannot be received from the switch. Refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting the ASAI digital link.
- -5 — Illegal request. The channel requesting the information is not a channel assigned by an ASAI domain to receive event messages. Make certain that you are using A_Event only in a non-voice script. Refer to Chapter 4, "ASAI Administration" for information.

When developing a program using Script Builder, A_Event returns an integer value in the Return Field to indicate the type of event (ABANDON, END, CONNECT, ROUTE REQUEST, or ABNORMAL ROUTE END) that is being reported. The possible values 65, 67, 69, 82, or 114 correspond to the ASCII representation of the first letter for each type of message.

For example, if an ABANDON is being reported, you may use the following evaluate statement:

Evaluate:

```
if ReturnCode = "A"  
is the same as  
if ReturnCode = 65
```

```
65 = "A" — ABANDON  
67 = "C" — CONNECT  
69 = "E" — END  
82 = "R" — ROUTE REQUEST  
114 = "I" — ABNORMAL ROUTE END
```

Table 6-2. A_Event Return Field Value Meaning

Return Value	Meaning	Explanation
65	ABANDON	The caller abandoned the call before the agent answered it.
67	CONNECT	If CONNECT is sent on alerting message, then this indicates that the agent specified in the Connected Number field has been alerted. If CONNECT is sent on connect message, then this indicates that the agent specified in the Connected Number field has answered the call.
69	END	The call ended. The caller has been disconnected or has hung up after the call was answered by an agent.
82	ROUTE REQUEST	The switch is requesting that the call be routed by the VIS. The number to be routed is in the Called Number field. The Routing ID field contains an identifier which must be used in the corresponding Routing ID field of the A_RouteSel action.
114	ABNORMAL ROUTE END	The switch indicates that the routing was unsuccessful. Either the Destination Number or the Split Extension specified in the A_RouteSel action is invalid or the call was abandoned before a route selection was made or vector processing was administered incorrectly. Check the Cause Value field for the specific reason.

Each ABANDON, CONNECT, END, ROUTE REQUEST, and ABNORMAL ROUTE END event returns information pertaining to the event. Table 6-3 indicates what fields the A_Event action returns for each event.

Table 6-3. Fields Returned by A_Event for Each Event

A_Event Field	Event		
	CONNECT ABANDON END	ROUTE REQUEST	ABNORMAL ROUTE END
Connected Party Number	X		
Calling Party Number	X	X	X
Called Party Number	X	X	X
Switch Data	X	X	X
Trunk Group Number	X	X	X
Call Id	X	X	X
Other Call Id	X	X	
LAI Display	X	X	
VIS Data	X		
Routing ID	†		

† The Routing ID field returns the ASAI cluster Id of the domain which reported these events.

Table 6-4. A_Event Fields

Field	Input/Output	Required?	Field Type	Field Size
Connected Number	output	required	char	20
Calling Party Number	output	required	char	20
Called Party Number	output	required	char	20
Switch Data	output	required	char	16
Trunk Group Id	output	required	num	–
Call Id	output	required	num	–
Other Call Id	output	required	num	–
LAI Display Info	ouput	required	char	15
VIS Data	output	required	char	20
Routing Id	output	reuiored	num	–
Cause Value	output	required	num	–
Return Field	ouput	optional	num	–

Defining A_RouteSel

The A_RouteSel action is used to send an ASAI route message to the PBX. The A_RouteSel action is used in conjunction with the A_Event action and may be used to construct scripts assigned to RTE type domains. Note that the A_RouteSel action should not be used in scripts assigned to Control (CTL), Automatic Call Distribution (ACD), and Vector Directory Number (VDN) type domains and may not be used in voice scripts.

To add A_RouteSel to a transaction while defining an application in Script Builder, press **ADD**. The Action Choices screen opens. Highlight “A_RouteSel,” then press **(ENTER)**. Press **CANCEL** to exit from the Action Choices screen.

You must now define the A_RouteSel step further. In the Define Transaction screen, highlight “External Action: A_RouteSel,” then press **DEFINE**. The Define A_RouteSel screen opens.

Figure 6-3. Define A_RouteSel Form Screen def_route.ps

Each of the fields in the Define A_RouteSel screen must contain a field name or constant which returns the following information. Refer to Table 6-5 for a summary of the information in each of these fields.



CAUTION:

You should specify only 'num' field types for those fields in the Define A_RouteSel screen that return numbers, that is, specify 'num' fields or constants for those fields that are of type 'num.'

The “Destination Number” field contains either a number or the name of a field which contains a number indicating where the call is to be routed. The Destination Number must be a valid extension number (for example, ACD hunt group extension, VDN, or station) or external address (for example, Direct Distance Dialing Number). The Destination Number may be up to 20 digits in length. If calls are routed to a number that is not on the switch, the Destination Number may contain a Trunk Access Code or Automatic Alternate Routing/Automatic Route Selection prefix digit.



NOTE:

A route request may be rejected by setting the field identified by the “Destination Number” field to the null string. In this case, the call will not be adjunct routed. Subsequent treatment for the call depends on what type of default vector processing has been administered on the switch.

The "Split Extension" field is used only for direct agent calls. This field contains either a number or the name of a field which contains a number identifying a valid ACD split. The Split Extension may be up to 5 digits in length. If the Split Extension field is used, the call is treated as a direct agent call and routed to the agent identified in the Destination Number field via the split identified in the Split Extension field. If the Split Extension field contains no digits (null value), then the call is treated as a normal call rather than a direct agent call. An empty split extension is entered by leaving the field blank as opposed to entering " ".

The "Priority Call" field indicates whether the call is to be delivered as a regular or priority call. This field must contain either "Yes" or "No" or a field name which may contain either of these two values. Enter "Yes" if you want the call delivered as a priority call. Enter "No" if you want the call delivered as a regular call. If "Yes" is not specified in this field, the call is delivered as a regular call.

The "Routing ID" field contains a unique number which identifies the call being routed. The Routing ID must match the Routing ID previously retrieved in the A_Event action.

The "Cause Value" field returns an error cause if the route select is not successful. Note that the error cause is returned if the Return Field contains a value of -3. Refer to VIS column in Table D-6 in Appendix D, "Troubleshooting ASAI" for a complete listing of values.

The "Return Field" contains a return code indicating whether the action was successful. If the A_RouteSel action is successful, it returns a number greater than or equal to zero. If A_RouteSel is unsuccessful, it returns one of the following values:

- -1 — A_RouteSel could not send the request to route the call to ASAI. Check the Message Log Report for system errors. Refer to Chapter 4, "Reports Administration" in the *CONVERSANT Voice Information System Operations* for information.
- -2 — A_RouteSel did not receive a response from the ASAI for the request to route the call. Check to see if the ASAI system is running.
- -4 — ASAI Link is down and route select information cannot be received from the switch. Refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting the ASAI digital link.
- -5 — Illegal request. The channel using A_RouteSel is not for a RTE domain. A_RouteSel is being used in a script that has not been assigned to an RTE domain. Refer to Chapter 4, "ASAI Administration" for information on assigning A_RouteSel to a domain.
- -6 — Switch did not respond after receiving the route select information.
- "53" — Bad Routing ID. The Routing ID specified in A_RouteSel is invalid. Check to make sure that the same Routing ID received from ROUTE REQUEST Event is used in the A_RouteSel action.

- -3 — The ASAI system could not route the call. Check the Cause Value field for information on why the call could not be routed.
- -11 — Destination Number exceeds 20 characters.
- -13 — Split Extension exceeds 5 characters.
- -15 — Routing Id is 0 or less.

Table 6-5. A_RouteSel Fields

Field	Input/Output	Required?	Field Type	Field Size
Destination Number	input	required	char	20
Split Extension	input	optional	char	5
Priority Call?	input	required	char	3
Routing ID	input	required	num	—
Cause Value	output	required	num	—
Return Field	output	optional	num	—

Defining A_Tran

The A_Tran action is used by voice scripts running on VIS agent (T/R or LST1) lines to transfer a call to a live ACD, live agent, or other station on the PBX. To do so, the A_Tran action takes control of the incoming call, puts the call on hold, places a call to the Destination Number and, if the call is not busy or denied, merges the original call with the call to the Destination Number. If the call is busy or denied, A_Tran drops the call to the Destination Number, reconnects to the original caller, and relinquishes control of the original incoming call.

⇒ NOTE:

This note is important when using A_Tran to transfer to VDNs/vectors. A_Tran merges the original incoming call with the second call only after determining that the second call has been placed successfully. The second call is considered to have been placed successfully if it becomes queued or alerts at an agent's telephone. Before the two calls are merged to complete the transfer, the original caller remains on hold. In some cases, you may wish to expose the caller to additional vector processing after the call is transferred (for example, play announcements or perform call prompting operations).

To insure that the original caller is exposed to such operations, you must construct your vectors so that the transfer is forced to complete before such operations are performed. Examples include queuing the call to the agent split before playing announcements and queuing the call to a dummy split before performing call prompting operations. No ports or stations need be dedicated to establish a dummy split. To insure that a "queue to" step will cause calls to be queued to a dummy split, however, the dummy split should be made vector controlled and queue slots should be assigned to it.

To add A_Tran to a transaction while defining an application in Script Builder, press **ADD**. The Action Choices screen opens. Highlight "A_Tran," then press **ENTER**. Press **CANCEL** to exit from the Action Choices screen. You must now define the A_Tran step further. In the Define Transaction screen, highlight "External Action: A_Tran," then press **DEFINE**. The Define A_Tran form screen opens (Figure 6-4).

Figure 6-4. Define A_Tran Form Screen def_tran.ps

Each of the fields in the Define A_Tran screen must contain a field name or constant which returns the following information. Refer to Table 6-6 for a summary of the information in each of these fields.

**CAUTION:**

You should specify only 'num' field types for those fields in the Define A_Tran screen that return numbers, that is, specify 'num' fields or constants for those fields that are of type 'num.'

The “Destination Number” field contains either a number or the name of a field which contains a number indicating where the call is to be transferred. The Destination Number must be a valid extension number (for example, ACD hunt group extension, VDN, or station) or external address (for example, Direct Distance Dialing Number). The Destination Number may be up to 20 digits in length. If calls will be transferred to a number that is not on the switch, the Destination Number may contain a Trunk Access Code or Automatic Alternate Routing/Automatic Route Selection.

The “Split Extension” field is used only for direct agent calls. This field contains either a number or the name of a field which contains a number identifying a valid ACD split. The Split Extension may be up to 5 digits in length. If the Split Extension field is used, the call is treated as a direct agent call and transferred to the agent identified in the Destination Number field via the split identified in the Split Extension field. If the Split Extension field contains no digits (null value), then the call is treated as a normal call rather than a direct agent call. An empty split extension is entered by leaving the field blank as opposed to entering “ ”.

The “Priority Call” field indicates whether the call is to be delivered as a regular or priority call. This field must contain either “Yes” or “No” or a field name which may contain either of these two values. Enter “Yes” if you want the call delivered as a priority call. Enter “No” if you want the call delivered as a regular call. If “Yes” is not specified in this field, the call is delivered as a regular call.

The “VIS Data” field contains up to 20 characters of any type of information that is saved and later used by the A_Event action. The data provided in this field is sent in the VIS Data field that appears in all messages received by the A_Event action for the transferred call, provided that the call is reported on either a CTL, VDN, or ACD type domain for the non-voice application.

The "Call State" field stores the status of the call. If a call transfer was attempted, the Call State field will have one of the following values:

- 0 — Call State information not available.
- 1 — Destination Number is ringing (alerting).
- 4 — Transfer was denied. Check the Cause Value field for additional information.
- 5 — Call to Destination Number is queued until a line becomes available.
- 7 — Destination Number is busy.
- 8 — The destination address seized a non-ISDN trunk (trunk seizure).
- 9 — The destination address is interworking with a non-ISDN trunk (cut-through).

**NOTE:**

The Call State field contains the status of the call when A_Tran returns 0, -53, -60, -61, and -62 in "Return Field".

The "Cause Value" field returns an error cause if the transfer is not successful. Note that the "Cause Value" field contains the reason for the failure if A_Tran returns -16, -17, -18, -19, or -20 in the Return Field. Otherwise, if A_Tran is successful, the Cause Value is set to -1. Refer to the VIS Value column in Table D-6 in Appendix D, "Troubleshooting ASAI" for a complete listing of Cause Values.

The "Return Field" returns a code indicating whether A_Tran was successful. A_Tran attempts to complete (merge) the transfer if the call is not busy or denied. If the A_Tran action is successful, it returns a number greater than or equal to zero. Otherwise A_Tran reconnects back to the original caller if necessary and returns one of the following negative values:

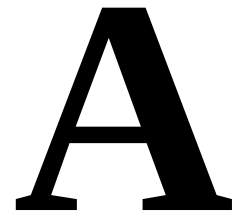
- -4 — ASAI link is down and transfer information cannot be performed. Refer to Appendix D, "Troubleshooting ASAI" for information on troubleshooting the ASAI digital link.
- -5 — Illegal request. The call is not being monitored so no transfer information is available. Refer to Chapter 4, "ASAI Administration" for information on Channel Administration and Domain Administration.
- -6 — Switch did not respond with the transfer information.
- -16 — A_Tran received an error from the ASAI system when trying to take control of the original call. Check the Cause Value field for information on why the request failed.
- -1 — A_Tran could not send the request to take control of the original call. Check the Message Log Report for system errors. Refer to Chapter 4, "Reports Administration" in the *CONVERSANT Voice Information System Operations* for information.

- -2 — A_Tran did not receive a response from the ASAI for the request to take control of the original call. Check to see if the ASAI system is running.
- -17 — A_Tran received an error from the ASAI system when trying to put the original call on hold. Check the Cause Value field for information on why the request failed.
- -18 — A_Tran received an error from the ASAI system when trying to place the call to the Destination Number. Check the Cause Value field for information on why the request failed.
- -20 — A_Tran dialed out to the destination number but the call was busy or denied. Check the Call State and Cause Value fields for the reason why the request failed.
- -19 — A_Tran received an error from the ASAI system when trying to complete (merge) the transfer. Check the Cause Value field for information on why the request failed.
- -11 — Destination Number exceeds 20 characters.
- -12 — Destination Number cannot be empty (0 characters in length).
- -13 — Split Extension exceeds 5 characters.
- -14 — VIS Data exceeds 20 characters.

Table 6-6. A_Tran Fields

Field	Input/Output	Required?	Field Type	Field Size
Destination Number	input	required	char	20
Split Extension	input	optional	char	5
Priority Call?	input	required	char	4
VIS Data	input	required	char	20
Call State	output	required	num	–
Cause Value	output	required	num	–
Return Field	output	option	num	–

Sample Scripts



Sample Scripts Overview

This chapter provides examples of scripts developed using the AT&T Adjunct/Switch Application Interface (ASAI) feature on the CONVERSANT Voice Information System (VIS). Included in this chapter is an ASAI voice script developed with the A_Callinfo and A_Tran actions, a routing script developed with the A_Event and A_RouteSel actions, and a monitoring script developed with the A_Event action.

Sample ASAI Voice Script

Following is an example of an ASAI voice script that has been developed with the A_Callinfo and A_Tran actions.

```
start:
# This is a sample voice script making use of the A_Tran action.
# This script would be used to handle calls at a T/R channel.
#
# In steps 1 through 3, standard Script Builder actions
# can be used to greet the caller, collect information, etc. In
# particular, it is assumed that a Prompt and Collect is used to
# collect an account number which is stored in account_num.
# A local database is read in an attempt to match the account
# number the caller provided and the ANI supplied with A_Callinfo.
# If a match is found, the table provides
# an agent extension and a split extension which are used to route the
# call to a specific agent within a split (direct agent routing).
# If no match is found, the call is routed to a default live agent
# split.
#
# Fields dest_num (agent extension) and split_num (split extension) for
# direct agent routing are returned from the table when a match is
# found.
#
4.  External Action: A_Callinfo
    calling: calling_num
    called: called_num
    switchdata: switch_data
    trunkid: trunk_num
    callid: call_id
    cause: callinfo_cause
    Return Field: callinfo_return
5.  Read Table
    Table Name: account_db Search From Beginning
    Field: account = account_num
    Field: ani = calling_num
#
# Set defaults in case no match is found in the table: dest_num is set
# to the default live agent split (split 5678). split_num is set to
# NULL so that direct agent calling is not invoked.
```

```
#
6. Evaluate
  If $MATCH_FOUND = 0
7.     Set Field Value
        Field: dest_num = "5678"
        Field: split_num = ""
    End Evaluate
#
# Transfer the call. Place the account number (account_num) in the
# visdata field. The ASAI DIP on the VIS saves this data and
# associates it with the transferred call. A subsequent CONNECT event
# reported for the transferred call will contain this data.
#
8. External Action: A_Tran
    destination: dest_num
    split: split_num
    priority: No
    visdata: account_num
    state: call_state
    cause: tran_cause
    Return Field: tran_return
#
# Note that the CONNECT event is not received in this voice script.
# Rather, a monitoring script is used to monitor the transferred
# call and receive the CONNECT event when the transferred
# call is delivered to an agent. This allows the T/R channel to service
# other calls while the first, transferred call is queued for an
# available agent.
#
9. Quit
```

Sample Routing Script

Following is an example of an ASAI routing script that has been developed with the A_Event and A_RouteSel actions.

```
start:
# This is a sample routing script making use of the A_Event action.
# This script would be given, via administration, a "RTE" type
# designation and therefore would receive only route requests (that is,
# no CONNECT, ABANDON, or END messages would be received or
# processed by this script). A local database is used to route the call based
# on ANI. A local database is read in an attempt to match the ANI
# for the call. If a match is found, the table provides an agent
# extension and a split extension which are used to route the call
# to a specific agent within a split (direct agent routing). If no
# match is found, the call is routed to a default split (for example,
# to a VIS T/R split to collect additional information).
#
# Fields dest_num (agent extension) and split_num (split extension) for
# direct agent routing are returned from the table when a match is
# found.
#
begin_loop:
#
1. External Action: A_Event
   connected: connect_num
   calling: calling_num
   called: called_num
   switchdata: switch_data
   trunkid: trunk_num
   callid: call_id
   otherid: other_id
   laidisplay: lai_info
   visdata: vis_data
   routingid: routing_id
   cause value: cause
   Return Field: event_return
#
# Check to make sure a ROUTE REQUEST was received.
# If a ROUTE REQUEST was not received, go back and get the next event.
#
2. Evaluate
   If event_return != R
3. Evaluate
   If event_return = r
```

```

4.          Modify Table
            Table Name : rtg_err Operation: Add
            Field: clg_num = calling_num
            Field: cld_num = called_num
            Field: err_cause = cause
            Field callid_value = call_id

#
#
#
#
            Else
5.          Goto begin_loop
            End Evaluate
        End Evaluate
6.  Read Table
            Table Name: ani_db Search From Beginning
            Field: ani = calling_num

#
# Set defaults in case no match is found in the table: dest_num is set
# to the default destination (split 1234). split_num is set to NULL
# so that direct agent calling is not invoked.
#
7.  Evaluate
    If $MATCH_FOUND = 0
8.      Set Field Value
            Field: dest_num = "1234"
            Field: split_num = ""
        End Evaluate
9.  External Action: A_Routesel
            destination: dest_num
            split: split_num
            priority: No
            routingid: routing_id
            cause: cause
            Return Field: route_return

#
# Repeat the process - go back and get the next event.
#
10. Goto begin_loop

```

Sample Monitoring Script

Following is an example of an ASAI monitoring script that has been developed with the A_Event action.

```
start:
# This is a sample monitoring script making use of the A_Event action.
# This script would be given, via administration, a "VDN",
# "ACD", or "CTL" type designation. This script would be used to
# receive information about monitored calls and pass this
# information to a host. In this type of scenario, the A_Event
# action can be used to receive CONNECT, ABANDON, and END events
# (no ROUTE REQUEST events are received). In this example, a subset of
# the information available in CONNECT events is passed to a host
# via the 3270 interface.
#
# It is assumed here that the Transaction Base Screen for the
# host application is called "connect_data". This screen is assumed
# to contain fields that are used for transmitting data obtained
# through A_Event. When the host receives the filled screen, it
# responds by sending a different screen, conveniently named
# the "verify" screen. The "verify" screen is then sent back
# with the key, PF3, to obtain the Transaction Base Screen,
# "connect_data", again.
#
begin_loop:
#
HOST_UP:
Event_start:
1. External Action: A_Event
   Connected_Number: connect_num
   Calling_Party_Number: calling_num
   Called_Party_Number: called_num
   Switch_Data: switch_data
   Trunk_Group_Id: trunk_id
   Call_Id: call_id
   Other_Call_Id: ocall_id
   LAI_Display_Info: lai_info
   VIS_Data: vis_data
   Routing_ID: route_id
   Return Field: event_ret
#
# Check to make sure a CONNECT was received since we don't care about
# ABANDON's and END's in this example application. If a CONNECT
# was not received, go back and get the next event.
#
```

```
2. Evaluate
    If event_ret != "C"
3. Goto Event_start
    End Evaluate
#
# Send data to the host. Only connected agent, ANI, DNIS, and
# VIS data are used in this example application.
#
# It is assumed that Aid Key for sending the data to the host is
# PF3. Note that you have to investigate what Aid Key is
# appropriate for your host environment.
#
8. Send Host Screen
Send Screen Name: connect_data Use Aid Key: PF3
    Field: connect_number = connect_num
    Field: ani = calling_num
    Field: dnis = called_num
voice_data = vis_data
9. Get Host Screen
For Screen Name: verify
End Get Host Screen
10. Send Host Screen
Send Screen Name: verify Use Aid Key: PF3
11. Get Host Screen
For Screen Name: connect_data
End Get Host Screen
#
# Repeat the process - go back and get the next event.
#
12. Goto Event_start
HOST_DOWN:
13. Goto start
```


ASAI Performance Performance Overview

This chapter contains performance information to be used with the AT&T Adjunct/Switch Application Interface feature. The topics included in this chapter are:

- Voice response integration performance
- Data screen delivery performance
- Routing application performance

For general information concerning performance of the CONVERSANT Voice Information System (VIS), refer to the Appendix E, "Performance Information" in *CONVERSANT VIS Version 4.0 Operations*, 585-350-703.

Each VIS can support one ASAI link. Up to eight VIS machines can be connected to the ISDN BRI Line Circuit Pack installed on the DEFINITY Generic 3i. The following gives the call volume capacities that each VIS can support based on the type of application. Note that the values vary depending upon whether you are using the local database or the remote database (accessed via the 3270 link), the number of records in the database, duration of the script being played, etc.

Voice Response Integration

Up to 2,500 busy hour calls are supported by the VIS. Voice Response Integration involves calls offered to VIS T/R or LST1 lines and transferred to an agent with data screen delivery.

Data Screen Delivery

Up to 10,000 busy hour calls are supported by the VIS. Data Screen Delivery involves calls offered directly to a monitored live agent split with either the VIS or the host delivering a data screen to the agent terminal. Note that this call limit assumes that the VIS delivers the screens.

Routing Applications

Up to 5,000 busy hour calls can be supported and 256 simultaneous Routing Requests can be handled by the VIS. Note that these call and request limits assume that the routing database is on the VIS machine.

In addition, note that a maximum of 48 channels can be administered on the VIS and a maximum of 32 domains (ACD & VDNs) can be monitored.

Note that each domain requires one virtual channel for running the script that is associated with it. With an increase in load, however, there might be a need for more virtual channels per script. Consequently, it is recommended that at least 2 virtual channels be reserved per script so as to insure reliable service.

As load increases on a domain, more instances of the script that is assigned to that domain are spawned off on the virtual channel. When the load is reduced, these virtual channels are released. However, at any time there is at least one script running per virtual channel. The number of virtual channels is set to 32 by the ASAI installation package.

It is recommended that you disable any unused domains so that the virtual channels associated with the domains may be used for other applications. Refer to Chapter 4, "ASAI Administration" for information on disabling the domain through the Domain Administration screen.