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Bell Labs Innovations

Intuity Conversant System V6.0

Communication Development

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Part 68: Answer-Supervision Signaling. Allowing this equipment to be operated in a manner that does not provide proper answer-supervision signaling is in violation of Part 68 Rules. This equipment returns answer-supervision signals to the public switched network when:

- Answered by the called station
- Answered by the attendant
- Routed to a recorded announcement that can be administered by the CPE user

This equipment returns answer-supervision signals on all DID calls forwarded back to the public switched telephone network. Permissible exceptions are:

- A call is unanswered
- A busy tone is received
- A reorder tone is received

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This document was prepared by the Product Documentation, Lucent Technologies, Columbus, OH.

RECYCLED PAPER

Contents

Contents

Contents

Contents

Contents

Contents

About This Book

Purpose

This book is a reference manual for creating the necessary platform environment and applications to implement various communication interfaces between callers, administrators, and the Lucent Intuity™ CONVERSANT® System.

How This Book Is Organized

This book is organized into the following sections:

- Chapter 1, “Intuity CONVERSANT System Communications Overview”

  This chapter provides a brief overview of the communications interfaces available within the system. This information includes both telephony and data network architectures.

- Chapter 2, “Analog Telephony Interfaces”

  This chapter describes the use of analog telephony as a communication arrangement, as well as the provisioning required to implement this interface. This includes the suggested administrative values to set on the system.

- Chapter 3, “Digital Telephony Interfaces”

  This chapter describes the use of digital telephony as a communication arrangement, as well as the provisioning required to implement this interface. This includes the suggested administrative values to set on the system.

- Chapter 4, “Adjunct/Switch Application Interface”

  This chapter describes the use of the Adjunct/Switch Application Interface (ASAI) as a communication arrangement, as well as the provisioning required to implement this interface. This includes the suggested analog or digital administrative values to set on the system.

- Chapter 5, “Converse Vector Step Routing”

  This chapter describes the use of the Converse Vector Step (CVS) routing as a communication arrangement, as well as the provisioning required to implement this interface. This includes the suggested administrative values to set on the system.

- Chapter 6, “Call Classification Analysis”

  This chapter describes the potential use and benefits of Call Classification Analysis (CCA) within analog and digital communication arrangements, as well as the provisioning required to implement this interface. This includes the suggested administrative values to set on the system.

- Chapter 7, “Data Network Communications”

  This chapter describes the potential uses of data network communications, discusses physical and logical protocol differences, and details what you must do on the system to implement this type of communication.

- Chapter 8, “Data Network Connectivity Alarms”

  This chapter describes the potential use of data network alarming and details what you must do on the system to implement this type of monitoring.

- Appendix A, “Transmission Level Adjustment”

  This chapter describes how to ensure that all speech heard by a caller is at a level which is appropriate for listening without causing oscillations or distortions in the network.

The book also includes a list of abbreviations, a glossary, and a cross-referenced index.

Conventions Used in This Book

The following typographic conventions are used in this book:

- Terminal keys
  - — Terminal keys are shown in rounded boxes. For example, an instruction to press the enter key is shown as

    Press ENTER.

- Function keys (also known as *soft* keys) are shown in rounded boxes followed by the function of that key in parentheses. For example, an instruction to press function key 2 is shown as

  Press F2 (CHOICES).

- Two or three keys that you press at the same time (that is, you hold down the first key while pressing the second and/or third key) are shown as a series of rounded boxes. For example, an instruction to press and hold ALT while typing the letter **d** is shown as

  Press ALT D.

- User input

  - The word *enter* means to type a value and press ENTER. For example, an instruction to type **y** and press ENTER is shown as

    Enter **y** to continue.

  - The word *type* means to press the key or sequence of keys specified. For example, an instruction to type **y** is shown as

    Type **y** to continue.

    Do *not* press ENTER after you type the value specified.

  - The word *select* is used to mean one of the following:

    a. Move to the desired menu item using the arrow keys and press ENTER. For example, an instruction to select an item from a menu and press ENTER is shown as

    Select Configuration Management from the Voice System Administration menu.

    b. Type the first character of the item. The first menu item beginning with that letter is selected. If more than one item begins with the same letter, then type enough letters to identify the desired item. Press ENTER when the correct item is highlighted.

  - Information that you enter or type from your terminal keyboard is shown in **bold** type; for example

    Enter **root** at the `Console Login` prompt.

  - Command and file names and their parameters are shown in **bold** type. Variable parameters are shown in ***bold italic*** type when they are part of a user input and in *regular italic* type when they are not. All are illustrated in the following example:

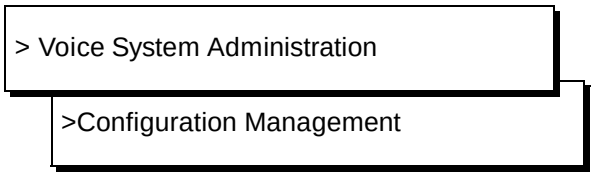
    Use the **print** command to print your report. The command syntax is **print *reportname***, where *reportname* is the name of the report to be printed.

- Screen displays
  - — Information that is displayed on your terminal screen — including screen displays, prompts, script code, and system messages — is shown in `typewriter-style` type; for example

    ```
    Installation is in progress -- do not remove
    the floppy disk.
    ```

  - — The sequence of menu options that you must select to display a specific screen is shown as follows:

    Begin at the CONVERSANT Administration menu, and select the following sequence:

    In this example, you would first access the CONVERSANT Administration menu. Then you would select the Voice System Administration option to display the Voice System Administration menu. From that menu, you would select the Configuration Management option to display the Configuration Management screen.

  - — The screens shown in the INTUITY CONVERSANT library are only examples. Your screens may not appear exactly as illustrated.

Related Resources

The following books are to be used in conjunction with the *Intuity* ™ CONVERSANT® System Version 6.0 *Communication Development* book:

- *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591 — This book describes the INTUITY CONVERSANT System screen-based administrative interface. It describes each screen in detail, including the various fields and their possible values. Refer to this book when it is necessary to perform administrative procedures for various communication scenarios.
- *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods* 585-310-761 — This book describes application development using TSM script language. Refer to this book for information related to TSM script language.
- *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760 — This book describes application development using the CONVERSANT Script Builder. Refer to this book for information related to Script Builder, such as external actions and external functions.

- Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *System Description*, 585-310-241, for a complete list of system documentation.

Technical Updates

Every effort was made to ensure that the information contained in these books is technically accurate, and will guide readers in the normal operation of the system. There are instances however, when the INTUITY CONVERSANT product may behave differently than is documented in the core library, or when hardware or software changes are made after these books have been published.

To help with this, an online bulletin board is available to all INTUITY CONVERSANT customers that provides supplemental information about this product in an electronic, E-mail format. These updates include hints, tips, and exception conditions about all aspects of the INTUITY CONVERSANT product that were discovered after the core library was published.

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INTUITY CONVERSANT System Communications Overview

1

What's in This Chapter

This chapter provides a brief overview of the communications interfaces available within the INTUITY CONVERSANT System. This information includes both telephony and data network architectures.

Communication Architecture

The INTUITY CONVERSANT System connects to the public switch telephone network (PSTN) to communicate with external callers. In some system applications, it also connects to private data networks to access host computer databases for information required to complete certain types of calls.

This book provides descriptions for the following types of interfaces:

- Telephony
  - — Analog
  - — Digital
- Data Network
  - — Asynchronous
  - — Synchronous

This book also provides information on communications specifications, connectivity, typical cabling arrangements, administration, and switch integration parameters, and discusses application development issues.

Public Switched Telephone Network

The INTUITY CONVERSANT interface to the PSTN uses either an analog connection or a digital connection. An analog transmission is a method of sending signals in which the transmitted signal is similar to the original signal. A digital transmission is a method sending information in which the transmitted signal is encoded as a series of ones and zeros.

The following features use either analog or digital connections:

- Adjunct/Switch Application Interface (ASAI)
- Converse Vector Step (CVS)
- Call Classification Analysis (CCA)

Chapters 2–6 should be read to learn more about the telephony interfaces used by the caller accessing the CONVERSANT System. Each of these chapters contain examples of how communications between the system and an external network are established. These examples are *not* the only methods of gaining this access as actual network cabling varies on a site-by-site basis. These chapters also provide examples of using various features in an application whether it was developed using Script Builder, transaction state machine (TSM) script language, or the INTUITY Response Application Programming Interface (IRAPI).

Private Data Network

The INTUITY CONVERSANT System supports two different forms of private data network interfaces: asynchronous and synchronous. These interfaces provide connections from the system to other computing devices such as remote monitoring systems or host computer databases. External customer interaction is not involved in these types of communications. Instead, the arrangement of these connections is based on the needs of the application being used. These private data networks are transparent to the caller who is invoking the application over the PSTN.

The following data network interfaces are supported and described in Chapter 7, "Data Network Communications" of this book:

- SNA 3270
- TCP/IP
- Token Ring
- SQL*NET
- Physical asynchronous connections to the CONVERSANT platforms

Various data network alarming packages are also available for use in conjunction with the INTUITY CONVERSANT System. The following packages are described in Chapter 8, "Data Network Connectivity Alarms"," of this book:

- NetView
- CompuLert/SCCS
- External Alarms

Analog Telephony Interfaces

2

What's in This Chapter

This chapter describes the Tip/Ring (T/R) and FAX analog telephony interfaces available with the INTUITY CONVERSANT System's base and optional software and the requirements that must be met to implement these interfaces.

This chapter also provides examples of typical analog connections.

Introduction to Analog Communications

In its analog configuration, the system provides nearly universal connectivity to existing private branch exchange (PBX) and automatic call distribution (ACD) premise equipment. It also allows standard interfaces to such widespread network services as Public Switched Telephone Networks (PSTNs) and Centrex service.

The following base analog telephony features make the INTUITY CONVERSANT System compatible with a variety of domestic PBXs or ACDs (including the Lucent DEFINITY Communications Systems Generic 1, 2, and 3, System 85, System 75, System 25, Dimension 2000, etc.):

- The system can perform switch-hook-flash transfers (also known as register recall) using the functions of the PBX or ACD or Centrex service. It can also determine if the extension to which the call was transferred is busy or there is no answer and whether an alternative message or action should occur.

- In addition to switch hook flash, the system supports transfer with a call bridge connection through the system. This bridging can be done with both digital and analog connections. Table 2-1 lists the analog line capabilities supported by call bridge.
- The system is capable of far-end caller disconnect detection through "wink signal" detection or such alternatives as call progress tone detection, for instance: dial tone, busy tone, or reorder tone detection. (The wink signal is a momentary break in loop current: typically 600 mSec.) Because these capabilities allow the system to know when a caller hangs up, the system rarely transfers a "ghost" call, but. instead allows the voice script to terminate quickly and to be ready for the next call.

  NOTE:
  Far-end caller disconnect detection through a wink signal or a call progress tone must also be supported by the PBX or ACD. Lucent DEFINITY Communications Systems Generic 1, 2, and 3, System 85, System 75, System 25, and Dimension 2000 provide the signaling needed to notify the CONVERSANT System of far-end caller disconnect. Other PBX systems may not. In these cases, implement script timeouts to ensure script termination.

- Outdialing for call transfer can be done with either touch tone or dial pulse (sometimes called decadic dialing or loop disconnect signalling).
- With custom software, the system can be programmed to transfer calls using dial access codes (rather than switch hook flash) to support PBXs that use this method of call transfer.

Trainable dial tone, software-settable switch hook flash duration, and wink signal duration also add to the system's flexibility

Table 2-1. Maximum Digital Trunks/Analog Lines Supported by Call Bridge

MAP/100, 100C		MAP/40	
Answer/Originate	**Outbound/Bridging**	**Answer/Originate**	**Outbound/Bridging**
60 digital T1 (linked)	60 digital T1	24 digital T1 (linked)	24 digital T1
36 analog T/R	36 analog T/R	24 analog T/R	24 analog T/R
72 analog T/R	48 digital T1	24 analog T/R	24 digital T1
96 digital T1	24 analog T/R	24 digital T1	24 analog T/R
96 digital T1	24 digital T1		

Table 2-2 details the maximum number of analog and digital lines supported without call bridging.

Table 2-2. Maximum Digital Trunks/Analog Lines Supported without Call Bridge

	MAP/100, 100C Answer/Originate	MAP/40 Answer/Originate
Analog	72	48
Digital	96	48

Analog Connections to a 5ESS

Analog lines from the local service provider supply the physical interface between the switch and the INTUITY CONVERSANT System. The lines should be configured as a standard 2500 analog set on the switch. Refer to Chapter 6, "Switch Interfaces," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for an extended list of tunable parameters available with the various switch integration packages.

Analog Connections to Lucent PBXs

Analog connections between an INTUITY CONVERSANT System and a PBX can be made to accommodate the needs for basic system connectivity. They can also be made to support optional feature packages that can make use of analog connections. Adjunct/Switch Application Interface (ASAI) is an example of a feature that relies on or uses T/R interfaces between the CONVERSANT System and the PBX.

The following settings and configuration data must be present on the PBX for analog T/R communication between the PBX and the CONVERSANT System. The INTUITY CONVERSANT System is designed to accommodate switch integration with Lucent System 75/DEFINITY switches as a default. Integration with other PBXs may require that you set specific switch integration values through the Voice System Administration menu. Refer to Chapter 6, "Switch Interfaces," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for an extended list of tunable parameters and valid values for various PBXs.

- The domestic PBX must provide analog service using CCITT (International Telephone and Telegraph Consultative Committee) and LSSGR (LATA Switching Systems Generic Requirements) standards. All analog station packs on DEFINITY switches and DIMENSION meet these standards. However, the LC03 circuit card on the DIMENSION and the SN229 circuit card on the System 85/G2 are *not* recommended for connection to the INTUITY CONVERSANT System.

- Each analog port on the switch must be configured to communicate as a standard 2500 analog set with the ability to transfer/conference calls. Each port requires a station number, an appropriate Class of Service (COS)/Class of Restriction (COR), and a hardware port location.

  NOTE:
  On DEFINITY G1/G3 switches, ports routed to the CONVERSANT System must not have data restrictions in the COR, and "redirect notification" must be set to "y" if the INTUITY CONVERSANT System will be transferring calls to ACD splits staffed by Auto Answer (zip tone) agents.

- The station numbers assigned to INTUITY CONVERSANT System ports must be valid entries in the system dial plan.
- If you are using a MERLIN LEGEND:
  - — All analog trunks receiving calls from and getting calls for the CONVERSANT System must provide reliable disconnect.
  - — All T/R lines originating from the MERLIN LEGEND switch connected to the CONVERSANT System must be setup in a MERLIN LEGEND calling group as type "Generic VMI."
  - — You must administer the lines connected to the system with "outside line" dial tone. Refer to "Inside Dial Tone" in the *MERLIN LEGEND Communications System Installation, Programming, and Maintenance* for additional information.

Analog Connections to Other Switches

The INTUITY CONVERSANT System can interface with other switches if differences in communication protocols and parameter settings are taken into account. The proper setting of these parameters on both the switch and the INTUITY CONVERSANT System is essential for establishing communications between the two devices. Refer to Chapter 6, "Switch Interfaces," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for an extended list of tunable parameters. For specific values for each parameter, contact your local technical support organization.

T/R Interface

The T/R interface is provided through an analog (loop-start) T/R circuit card, with multiple 2-wire interfaces to the PBX, ACD, central office (CO), or foreign PSTN services. In addition to providing a physical network interface, the T/R circuit card provides speech encoding and playback, dual tone multifrequency (DTMF) recognition, call supervision, and intraswitch call classification for intelligent transfers. Refer to "Introduction to Analog Communications" for additional information.

T/R Connectivity

Refer to the hardware installation book for your platform for information on installing a T/R circuit card. Figures 2-1 through 2-3 show typical T/R connections from the INTUITY CONVERSANT System.

NOTE:
The connectivity diagrams provide examples of T/R connections and are not the only method(s) of gaining connectivity to an external network. Actual network cabling varies on a site-by-site basis, and the cabling techniques used in each installation are the responsibility of the system administrator or installation technician.

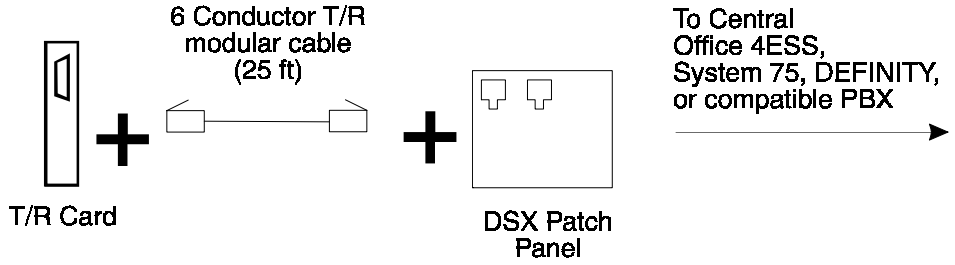
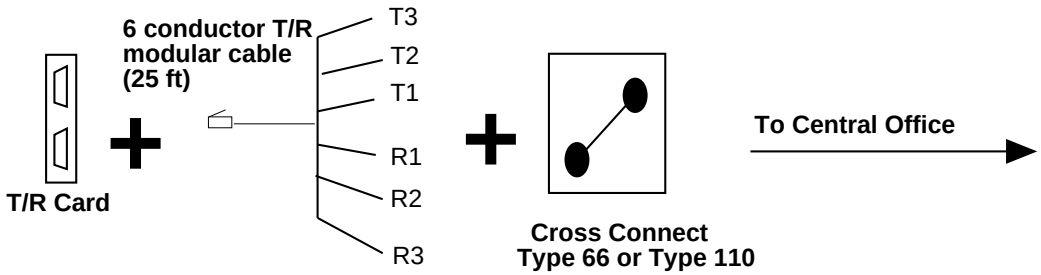


Figure 2-1. Analog T/R Interface Connection to a DSX Patch Panel



The AYC30 and AYC16 cards have 8-pin jacks where the outside pins are available for a rarely used “Earth Recall” feature.

Figure 2-2. Analog T/R Interface Connection to a Type 66 or 110 Cross-Connect

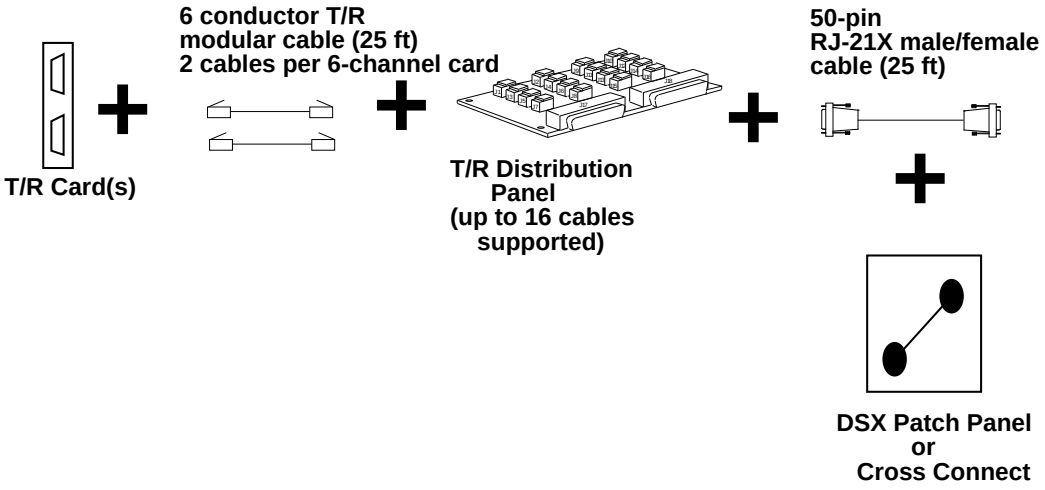


Figure 2-3. Analog T/R Interface Connection from Distribution Panel Using RJ21X Cable

T/R Telephony Interface Specifications

Tables from Table 2-3 through Table 2-7 detail the various T/R telephony interface specifications.

Table 2-3. T/R Circuit Card General Specifications

Attribute	Value
Type of service	Loop-start POTS
Loop current detection	15 mA minimum
Ringing voltage detection	88 VRMS at 20 Hz (nominal)
Ringer equivalence for T/R	0.5-0.8 B‡
Wink detection*†	80 – 800 msec
Flash duration*†	40 – 1550 msec
Register recall*	Timed break
Answer delay*	0 – 10 rings

*These attributes are adjustable through the Analog Switch Interface (ASI) packages.

†These attributes can be changed via the Analog Interfaces screen described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration,* 585-310-591.

‡ Varies with hardware type. See *Intuity* ™ CONVERSANT® System Version 6.0 *System Description* 585-310-241 for particulars.

Table 2-4. T/R Circuit Card DTMF Tone-Detection Specifications

Attribute	Value
Digits	0 – 9, asterisk (*), pound sign (#), A – D
Amplitude	+3 to -22 dBm total power (nominal tones)
On/Off timing	80 msec minimum on, 23 msec off
Gaps bridged	23 msec
Signal/noise ratio	23 dB (nominal tones at -19 dBm total power)
Twist	+4 to -8 dB (high to low tone)
Frequency deviation	+/-1.5 %

Table 2-5. T/R Circuit Card DTMF Addressing Specifications

Attribute	Default Value
Digits	0 – 9, asterisk (*), pound sign (#), A – D
On/off timing*	100 msec on, 60 msec off
Frequency	Precise tones
Twist*	0 dB
Amplitude*	-3 dBm per frequency
Dial Pulse Addressing Specifications	
Break Time*	60 mSec
Make Time*	40 mSec
Interdigit Time*	600 mSec

*These attributes are adjustable through the Analog Switch Interface (ASI) packages.

Table 2-6.

Table 2-7. T/R Circuit Card Default Call Progress Tone Detection Specifications

Tone	Frequency (Hz)*	Amplitude (dBm)*	S/N Ratio (dB)	Maximum Twist (dB)	Frequency Deviation (%)	Cadence*
Dial	350 + 440 †	+1 to -24	55	+3	+/-0.5	Present for 1 sec
Stutter dial (recall dial tone)	350 + 440 †	+1 to -24	55	+3	+/-0.5	3 cycles of 120–150 msec on, 120–150 msec off followed by 1 sec on
Ringback	440 + 480	+1 to -24	55	+3	+/-0.5	500–2000 msec on
Busy	480 + 620	+1 to -24	55	+3	+/-0.5	60 IPM, 250–500 msec on, 500–750 msec off
Reorder (Fast busy)	480 + 620	+1 to -24	55	+3	+/-0.5	120 IPM, 100–250 msec on, 250–450 msec off

T/R Circuit Card Administration

Placing a card in the INSERV state allows it to be used for the purpose (play, code, etc.) for which it is allocated in the application. You may need to *manually* place a T/R card into service if:

- After first installing the card or changing switch integration parameters, the voice system did not automatically place the card in the INSERV state
- The card was placed in the MANOOS state
- A diagnostic procedure failed (that is, placed that card in the MANOOS or BROKEN state)

To change the state of the T/R cards to INSERV, use the steps described in Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

Transmission Level Plan

A Transmission Level Plan (TLP) for a piece of telecommunications equipment is a set of specifications dictating the incoming/outgoing speech volume levels that pass through the equipment and the hardware and software tools for implementing those specifications. The specifications take into account the level plans of the various telephone network interfaces to which the equipment will connect. The goal of the plan is to ensure that all speech heard by a caller be at a level which is appropriate for listening without causing oscillations or distortions in the network.

Figure 2-4 shows most switch designs implement a TLP with a “built-in” gain of -3 dB (often called insertion loss) in each T/R loop of a station-set-to-station-set connection, for a total gain of -6 dB from end to end. The INTUITY CONVERSANT System default TLP implements this same strategy; that is, the system default TLP attempts to make the end-to-end gain of voice signals passing through it equal to -6 dB. (There are reasons to implement other strategies, however, see "Reasons for Deviating from the Default IVOL and OVOL Settings" on page A-5.)

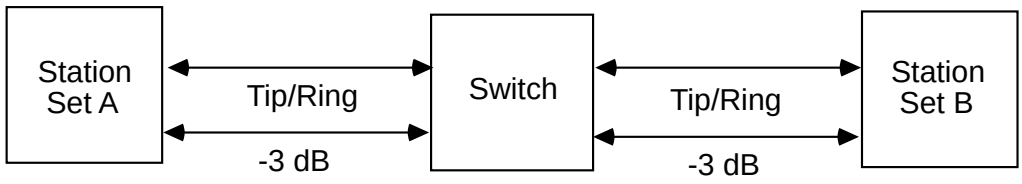


Figure 2-4. Typical Switch Transmission Level Plan for Station-Set-to-Station-Set Connection

Table 2-8. T/R Circuit Card Transmission Level Plan

Attribute	Value
Input gain	0 dB fixed
Output gain	0 dB fixed
Incoming speech volume (IVOL) – card voice coding only	Selectable from -9 to +12 dB
Outgoing speech volume (OVOL) – card voice playback only	Selectable from -9 to +12 dB
TDM output gains	Selectable from -30 to +6 dB

NOTE:
The IVOL, OVOL, and TDM output gains are system-wide parameters for analog interfaces and can be changed on a per-card basis for digital interfaces. These parameters can be modified via the Switch Interfaces screens as described in Chapter 6, "Switch Interfaces," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. Gains can also be overridden on a per-channel basis by an INTUITY Response Application Programming Interface (IRAPI) application. However, even with IRAPI, the IVOL cannot be overridden for speech recording on a T/R channel. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761, for the IRP_PLAYGAIN and IRP_RECORD_GAIN parameters under IrPARAMETERS(4IRAPI).

Refer to Appendix A for more information about adjustment of IVOL and OVOL levels.

FAX Interface

Facsimile (FAX) communications involve transmitting graphic and text images between FAX machines and other devices via standard telecommunications networks.

For a general discussion of the Script Builder FAX Actions, refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *System Description*, 585-310-241.

FAX Provisioning

The INTUITY CONVERSANT System supports a maximum of 12 FAX channels through the use of the Brooktrout TR114 and TR114+ (both are four-channel circuit cards). Up to three circuit cards may be used. Applications that use the Script Builder FAX Actions can be assigned to T/R, T1, or Line Side (LST1/LSE1) channels.

Refer to the hardware installation book for your platform for information on installing FAX circuit cards in the CONVERSANT System.

FAX Application Development Issues

The INTUITY CONVERSANT System can invoke FAX services through Script Builder applications.

Script Builder FAX Actions

The Script Builder FAX Actions allow you to include FAX communications in any Script Builder application. Script Builder FAX Actions offer the following capabilities:

- Transmit a prestored graphic image to a caller
- Transmit a dynamically created text image (file) to a caller
- Create a text file dynamically for transmission to the caller
- Create customized cover pages

For a general discussion of the FAX Actions, including benefits and uses, refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *System Description,* 585-310-241. For detailed information about implementing Script Builder FAX Actions in INTUITY CONVERSANT System applications, refer to Chapter 6, “Using Optional Features with Script Builder,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760.

For information about implementing FAX Actions by means of Graphical Designer, refer to *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Graphical Designer* 585-310-764.

Digital Telephony Interfaces

3

What's in This Chapter

This chapter describes the T1, Line Side T1 (LST1), E1, Line Side E1 (LSE1), and Primary Rate Interface (PRI) digital telephony interfaces available with the INTUITY CONVERSANT System base. It also describes optional software and the requirements that must be met to implement these interfaces.

This chapter also provides examples of typical digital connections, and discusses application development issues you must address when using the various digital telephony interfaces and their parameters.

Introduction to Digital Communications

A digital T1 (E&M) or E1 (CAS) circuit (trunk) allows the system to connect to digital network facilities such as a central office (CO) switch. Digital connections between a DEFINITY switch and the Conversant System can be through PRI, T1 (E&M), E1 (CAS), line-side T1, or line-side E1. (Generally either E1 or T1 service prevails in a given area and a choice between the two follows.)

Advantages of Digital Service

Analog configurations require one analog connection between the CONVERSANT System and a connected switch, for each incoming channel whereas several digital channels can be transmitted over a single connection. E1 CAS or LSE1 requires only one cable to provide 30 channels of service. T1 E&M or LST1 requires only one cable to provide 24 channels of service.

Digital connections also significantly reduce the number of circuit cards required to support a CONVERSANT-to-switch interface. Analog configurations require five IVP6 circuit cards to support 30 incoming channels. E1 or LSE1 reduces the required hardware to only one E1 circuit card (and part of an SP or SSP card). T1 or LST1 requires one T1 circuit card (and part of an SP or SSP card) per 24 channels of digital service. Two T1 circuit cards and one SP circuit card provides 48 voice channels.

The AYC3B and AYC11 circuit cards are used only for T1 services.The AYC21 card may be used for either E1 or T1 services.

Advantages of Primary Rate Interface (PRI)

PRI acts as a powerful interface between intelligent equipment such as PBXs and computers. Furthermore, PRI is widely used for access to features provided over the larger network such as automatic number identification (ANI).

See "Primary Rate Interface" on page 3-18 for a much more detailed discussion of features that accompany the use of PRI.

(PRI may be carried on either T1 or E1 lines. It provides 23 bearer (B) channels when carried over T1 lines, or 30 B channels when carried over E1 lines. In either event the calls are controlled from endpoint to endpoint by messages transferred over data (D) channels.)

Network Communications

A T1 digital circuit carries information at 1.544 Mbps, and consists of 24 DS-0 channels. Each DS-0 channel operates at 64 Kbps, and is the equivalent of one incoming data line. An AYC21 interface card has a (mechanical) switch by means of which one can choose to use either the T1 or E1 interface. The E1 interface is very similar to the T1 except that an E1 digital circuit carries information at a rate of 2.048 Mbps and consists of 30 B channels and two signalling and framing channels. Each B channel is the equivalent of one incoming data line. T1 and E1 tend to be alternatives used in differing locations.

T1 connections also provide dialed number identification service (DNIS) information to further automate incoming calls for customers with multiple 800 or 900 numbers. Table 2-2 on page 2-3 shows the maximum number of digital lines that are supported on each of the INTUITY CONVERSANT platforms.

T1, E1, and Integrated Services Digital Network (ISDN) PRI support trunk interfaces. ISDN PRI can operate at either the T1 or E1 rate. A T1-PRI interface contains either 23B+D channels or 24B channels that are associated with the D channel on another 23B+D card. An E1-PRI interface contains 30B+D channels. Currently E1 PRI is only supported when used in conjunction with an ACULAB protocol converter card. The D channel does not provide normal telephony

service, but is used to control the calls on the B channels. It provides information such as DNIS and ANI. Each B channel provides a 64 Kbps voice path.

Interconnection with PBX

Line-side connections between a DEFINITY switch and a Conversant System may be either by means of a line-side T1 (LST1) interface or by a line-side E1 (LSE1) interface (the interfaces just described). LST1 connections to a Galaxy ACD are also supported. Refer to "Line Side Digital Interface" on page 3-13 for more information.

These LST1 and LSE1 channels also support the Adjunct/Switch Application Interface (ASAI) feature when used with DEFINITY switches. (ASAI can be used for more advanced call control and to collect such information as ANI and DNIS.) See "Advantages of Using the ASAI Feature" on page 4-1.

T1 (E&M), E1 (CAS), and PRI connections to a DEFINITY switch are supported as well as LST1 and LSE1, but LST1 and LSE1 are generally preferable. LST1 and LSE1 support flash transfers but T1 (E&M), E1 (CAS), and PRI do not.

The system supports call bridging through a digital connection. Call bridges can also be used to simulate a transfer, but this consumes channel resources. Table 2-1 on page 2-2 lists the digital line capabilities that call bridge supports.

Digital Telephony Interface Specifications

Table 3-1 details the general digital telephony interface specifications for all T1/E1 protocols.

Table 3-1. Digital Telephony Interface General Specifications

Attribute	AYC11 & AYC3B	AYC21
Physical connector	Subminiature DB-15 male receptacle	BNC co-ax or 8-pin modular
FCC registration	AS593M-17926-VM-E	Pending
Safety approval	UL Type Approved	Under test
T1 Signal regeneration	CSU required over 200 meters (655 feet.)	
T1 Loopback capability	CSU required for remote capability	
Transmission Level Point (TLP) at DS-1 interface	0 ELP, 0 DLP	
TLP at time-division multiplexed (TDM) interface	0 ELP, 0 DLP	

Continued on next page

Table 3-1. Digital Telephony Interface General Specifications

Attribute	AYC11 & AYC3B	AYC21
Call progress tone frequency	Precise tone frequencies	Precise tone frequencies can be tuned to accomodate local standards
Call progress tone levels	-6 dBm total (nominal)	Same, but can be changed through digital switch interface packages.
Call progress tone timing	Ringing - on/off: 2 sec on, 4 sec off Busy - on/off: 0.5 sec on, 0.5 sec off	Same, but is tunable through digital switch interface packages.
Call progress tone detection	Not supported; must use optional CCA feature if this capability is required. [Note: Even with Full Call Classification Analysis (CCA), LST1 does not detect dial tone.]	Supported with Line Side Definity protocol (either T1 or E1transmission rate)
DS-1 timing source	Slave to DS-1 source (loop timed)	
DS-1 timing (free running)	Stratum 4	
Suggested channel service unit (CSU) types for use at T1 rate	Paradyne (PEC 21581-ESF), Verilink 551VST List 2, or equivalent	
Supported configurations	Tie trunk (robbed-bit E&M), ISDN-PRI, LST1	Tie trunk (robbed-bit E&M),E1 (CAS), ISDN-PRI (E1/T1), LSE1, LST1
Dual tone multifrequency (DTMF) output timing	70 msec on, 70 msec off	Same, but is tunable through digital switch interface packages.
DTMF output levels	-8 dBm per frequency (nominal)	Same, but is tunable through digital switch interface packages.

Continued on next page

Table 3-1. Digital Telephony Interface General Specifications

Attribute	AYC11 & AYC3B	AYC21
DTMF receivers	LATA Switching Systems Generic Requirements (LSSGR) compatible (Note: If DTMF muting is on for a call, the DTMF receiver's minimum on time for detection is increased and may not meet LSSGR requirements, DTMF muting does not impact LSSGR compatibility of DTMF receivers during call setup (that is, S digits).	Same, but is tunable through digital switch interface packages.
Number of receivers: T1	24 (one per DS-0 channel)	
Number of receivers: E1	E1 not supported on AYC11 or AYC3B	30 (one per B-channel)

Digital Connectivity

The MAP/100 and MAP/100C support up to five T1 circuit cards. The MAP/40 also supports five T1 circuit cards. A (Speech and) Signal Processor [(S)SP] circuit card is required if you are using T1 circuit cards in coding and playback situations.

NOTE:
Each SP circuit card supports up to 31 channels of simultaneous speech playback using adaptive differential pulse code modulation (ADPCM) 32-Kbps coded speech. Applications that require large amounts (more than 30 channels) of simultaneous speech coding may require additional SP circuit cards.

This precaution applies to a lesser degree to the SSP card since it supports up to 120 channels of simultaneous speech ADPCM playback.

Refer to the hardware installation book for your platform for information on installing Digital and (S)SP circuit cards.

The illustrations that follow show examples of typical digital connections to trunks and switches. Table 3-3 details the digital telephony specification for the T1.5 Robbed-bit E&M protocol. Use Table 3-3 in conjunction with Table 3-1.

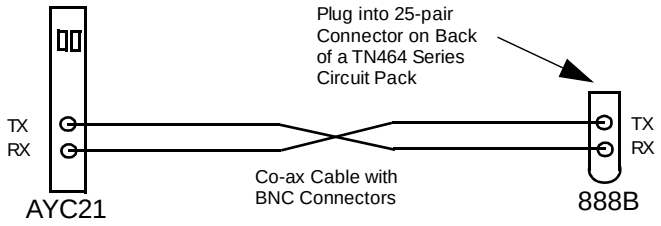


Figure 3-1. Example of AYC21 Co-axial Connections to a DEFINITY G3 Switch

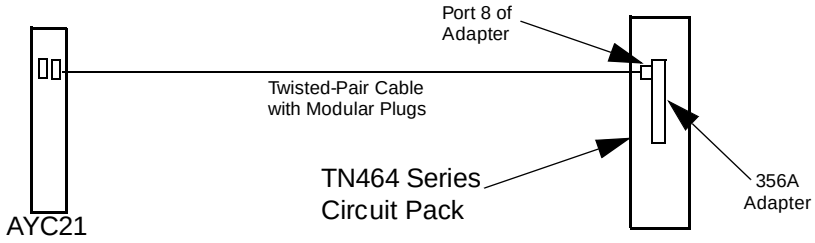


Figure 3-2. Example of AYC21 Twisted-Pair Connections to a DEFINITY G3 Switch

Channel Service Unit Connectivity (T1 Only)

The T1 interface circuit card is connected to a channel service unit (CSU) or directly to the DS-1 terminal block to establish T1 connections to a CO.

A CSU performs certain line-conditioning and equalization functions and responds to loopback commands from the CO. A CSU regenerates digital signals, monitors them for problems, and provides a way to test the digital circuit. A CSU is not always needed. However, a CSU is *required* if any of the following situations applies to the system setup:

- The INTUITY CONVERSANT System is more than 200meters (655 feet) from the signal source. The signal source may be a DSX or the last T1 repeater. Here, the CSU regenerates the received signal and properly attenuates the transmitted signal to prevent crosstalk.
- The INTUITY CONVERSANT System is terminating the T1 trunk from outside the building. Here, the CSU provides the primary lightning and surge protection as required by FCC Part 68.

- The T1 loop is not dry (that is, the loop is powered by either 110 VAC, +24 VDC or -48 VDC sources).
- You want to use the remote loopback and/or extended super frame (ESF) maintenance features Here, the CSU recognizes the in-band bit patterns that signal it to loopback the incoming signal or to perform other maintenance functions.

On some types of CSUs, the connector on the T1 cable can plug into the AYC3B, AYC11, or AYC21 circuit card and the cable terminates at a 15-pin D subminiature connector to the CSU.

On other types, you must cut off the CSU connector and slide latch and strip and connect the wires. AYC21 connection is given in Figure 3-6. The information given below applies to cables used to connect to AYC11 and AYC3B cards:

- Orange = our T1 = signals to CONVERSANT and should connect to a CSU or Network “T”
- White/Orange = our R1 = signals to CONVERSANT and should connect to a CSU or Network “R”
- Green = our T = signals from CONVERSANT and should connect to a CSU or Network “T1”
- White/Green = our R= signals from CONVERSANT and should connect to a CSU or Network “R1”

Refer to Figure 3-3, Figure 3-4, Figure 3-5, and Figure 3-6 for CSU connection examples.

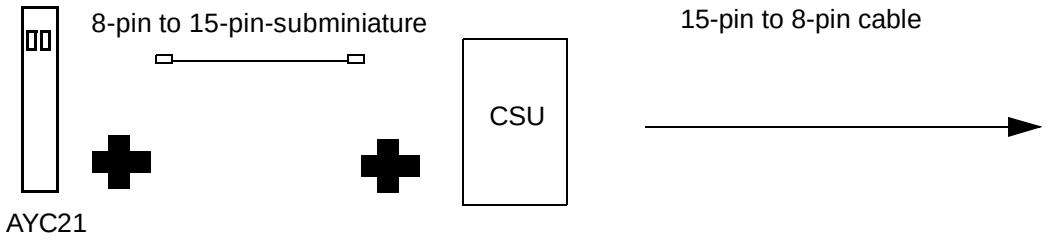


Figure 3-3. Example of T1 Interface Connection to a CSU (From an AYC21)

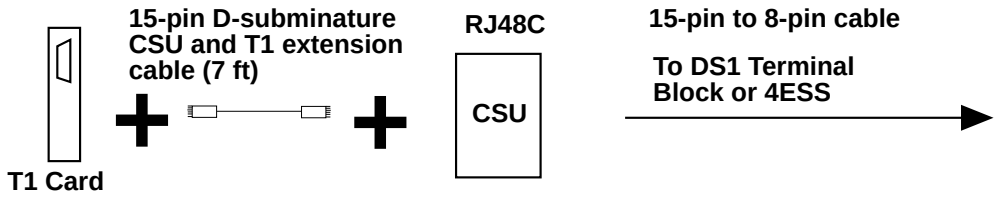


Figure 3-4. AYC11 or AYC3B Connection to a CSU with a 15-Pin Connector

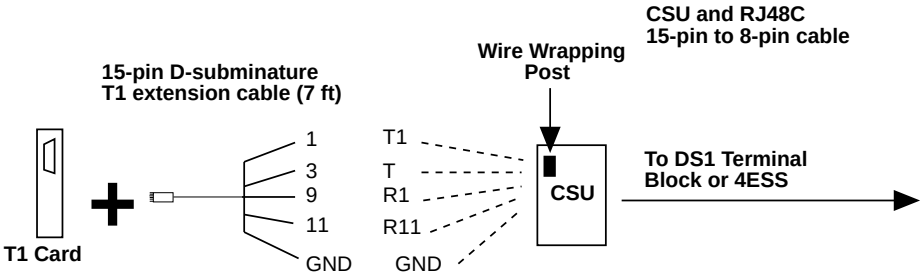


Figure 3-5. Example of AYC11 or AYC3B Connection to a CSU with Wire Wrapping Posts

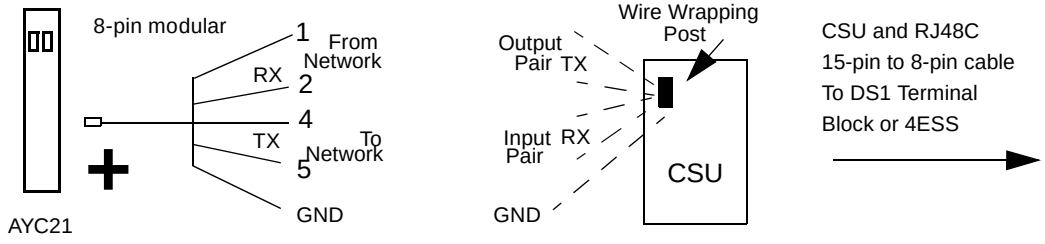


Figure 3-6. Example of AYC21 Connection to a CSU with Wire Wrapping Posts

E1 CAS (Channel Associated Signaling) Interface

The AYC21 circuit cards can operate with Channel Associated Signaling. This interface (at the E1 rate: 2.048 Mbits/sec) uses signaling bits associated with each channel to determine the state of the channel. Thirty voice channels are supported on each link.

Several country specific signaling protocols have been developed using E1-CAS. Contact your technical representive for more information about locally supported protocols.

Table 3-2. Digital Telephony Interface Specifications for E1 CAS Configurations.

ATTRIBUTE	SPECFIFICATION
DS1 Rate	2.048 Mbits/sec. (ITU G.703)
DS1 framing/line coding	HDB3 (ITU G.704, G.705)
Cyclic Redundancy Check (CRC)	(ITU G.706) May be set to YES or NO. Must match the CRC setting of the Network Entity connected to the AYC21.
Physical Connector Options:	120 Ohm twisted pair on RJ-48C modular jack, OR 75 Ohm BNC jacks.
PCM Companding Rule:	A-Law or Mu-Law. (ITU G.711)
Line Signaling	ITU System R2, Q.421 Compliant; Variations by specific protocol are supported.
Address Signaling Options (Register Signaling) Incoming and Outgoing	DTMF (Touch Tone) ITU System MFC, Q.440, Q.441; Variations for specific protocols are supported by table entries. Dial Pulse (Slower than DTMF or MFC)
Outgoing Destination Number:	15 digits Max.
Outgoing ANI Number:	15 digits Max. (if supported by protocol)
Incoming Address: (DNIS)	15 digits Max.
Incoming ANI Number:	15 digits Max. (if supported by protocol)

Continued on next page

Table 3-2. Digital Telephony Interface Specifications for E1 CAS Configurations.

ATTRIBUTE	SPECFIFICATION
Audible Alerting Tones on Incoming Calls	Ring, Busy, Reorder. Variations by country supported.
Call Progress Tone Recognition on Outbound Calls	Not supported.
Call Transfer Capability	Not supported.

E1 Switch Integration and Administration

Switch Integration for E1-CAS is done using the Digital Interfaces screen. This screen is described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. You must select one of the E1-CAS protocols that correspond to optional packages loaded on the INTUITY CONVERSANT.

Placing a card in the INSERV state allows it to be used for the purpose for which it is allocated in the application. After performing switch integration on the E1 circuit card for the CAS protocol, you may need to *manually* place a E1 circuit card into service if:

- After first installing the card or changing switch integration parameters, the voice system did not automatically place the card in the INSERV state
- The card was placed in the MANOOS state
- A diagnostic procedure failed (that is, placed that card in the MANOOS or BROKEN state)

To change the state of the E1 circuit cards to INSERV, use the steps described in Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

E1 Connections:

Because telephone network connections vary from country to country, no specific recommendation can be made concerning connection to the Network Entity. Please consult your local Lucent Technologies technical representative to determine the proper physical connectivity.

T1 E&M Interface

The T1 circuit cards accept an ISDN PRI or DS-1 two-way digital trunk and convert it to two-way analog audio channels. Because of bandwidth and transmission differences of each trunk, ISDN PRI and DS-1 offer different numbers of converted channels. A standard 1.544-Mbps DS-1 format trunk converts to 24 DS-0 channels. These 64-Kbps channels can provide 24 two-way audio channels.

Table 3-3. Digital Telephony Interface Specifications for T1 E&M Type Configurations

Attribute	Specification
DS-1 framing	D4 type only
DS-1 line coding	Zero code suppression (ZCS)
Protocol	Robbed-bit (4-wire) E&M
Alerting in/out	Wink/wink
Wink generation	230 msec default (Selectable: 20 – 2500 msec)
Wink detection range	100 – 350 msec
Addressing (outgoing)	DTMF (touch tone) MF (Multi-Frequency) Dial Pulse (slower than DTMF or MF)
Number of digits	15-digit maximum
Addressing (incoming)	DTMF (touch tone) MF (Multi-Frequency) Dial Pulse (slower than DTMF or MF)
Number of digits (DNIS)	Will wait for up to 16 digits (selectable). (This specification can also be provisioned not to wait for digits.)
Initial digit timer	Will wait up to 4 seconds for first digit. (This specification can also be provisioned not to wait for digits.)
Interdigital timer	Will wait up to 2 sec between digits
Audible ring starts	As soon as the selected number of digits is received or when one of the above timers expire, whichever occurs first

Continued on next page

Table 3-3. Digital Telephony Interface Specifications for T1 E&M Type Configurations

Attribute	Specification
DNIS capacity	0 – 16 digits
ANI capacity	Not supported
Transfer capability	Not supported

T1 Switch Integration and Administration

Switch integration for T1 is done using the Digital Interfaces screen. This screen is described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. You must select T1 A/B Robbed-bit E&M Protocol from the Digital Interfaces screen. (Refer to "Switch Integration and Administration" on page 3-15 and "Primary Rate Interface" on page 3-18 for information on performing switch integration for those types of protocols.)

Placing a card in the INSERV state allows it to be used for the purpose for which it is allocated in the application. After performing switch integration on the T1 circuit card for the E&M protocol, you may need to *manually* place a T1 circuit card into service if:

- After first installing the card or changing switch integration parameters, the voice system did not automatically place the card in the INSERV state
- The card was placed in the MANOOS state
- A diagnostic procedure failed (that is, placed that card in the MANOOS or BROKEN state)

To change the state of the T1 circuit cards to INSERV, use the steps described in Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

T1 Connections

T1 connections through channel service units are covered under "Channel Service Unit Connectivity (T1 Only)" on page 3-6.

Digital Application Development Issues

The AYC21 circuit card recognizes call progress tones and therefore supports flash transfers over line-side E1 or line-side T1. The AYC21 circuit card does not, however, support flash transfer over T1 (E&M), E1 (CAS), or PRI. The AYC11 and AYC3B circuit cards only support flash transfers when LST1 is accompanied by Full CCA, refer to "Line Side Digital Interface" on page 3-13.

Simulated transfers using digital cards can be performed over call bridges. In the analog Tip/Ring (T/R) or line-side digital environment, the switch-hook-flash transfer releases the call from the INTUITY CONVERSANT System once the transfer is made. A call bridge, however, ties up an incoming port and an outgoing port until the call has concluded. Thus, with two ports being tied up simultaneously, more digital ports may be necessary.

Script Language

The **tic** instruction is used for basic control of incoming and outgoing calls on T1 and E1 lines. For additional information about using the transaction state machine (TSM) script language on T1 lines, refer to the **tic** instruction in Chapter 3, “Script Instructions,” and Appendix B, “Summary of Script Instructions,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761.

INTUITY Response Application Programming Interface

The **irCall()**, **irAnswer()**, **irDial()**, and **irDisconnect()** functions provide the basic call control capabilities for T1 interfaces with the INTUITY Response Application Programming Interface (IRAPI). The **irStartSpeechED()** function is not supported for T1 interfaces over AYC11 or AYC3B cards. The **irStartSpeechED()** function is supported, however for LST1 or LSE1 interfaces over an AYC21 board. Refer to Chapter 6 of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761, for more information about these functions when developing IRAPI applications.

Line Side Digital Interface

Line Side T1 (LST1) allows the use of a 24-channel, 1.544-Mbps digital interface between a customer PBX and the INTUITY CONVERSANT V6.0 platform. LST1 uses T1 card technology with special protocol-level software and CONVERSANT System user interface modifications to improve system connectivity and reduce the number of circuit cards and cables required (relative to tip/ring technology) to support 24 channels of service. Line Side E1 (LSE1) allows the use of a 30-

channel, 2.048-Mbps digital interface between a DEFINITY G3 PBX and the INTUITY CONVERSANT System V6.0 platform. LSE1 provides a similar improvement in connectivity relative to tip/ring cards.

LST1 is compatible with DEFINITY G3 PBX and Galaxy 8 Automatic Call Distributing (ACD) systems. LST1 also supports the ASAI feature when used with DEFINITY G3 PBX. LSE1 is only supported for the DEFINITY G3 PBX. LSE1 also supports the ASAI feature.

Line Side E1/T1 Provisioning

When either LSE1 or LST1 is used to provide an ASAI link between the CONVERSANT System and a PBX a separate path must be provided for communications between the two systems. The path may be provided by either of a DEFINITY LAN Gateway connected to a local area network, or a BRI D-channel link over an IPCI card.

The following limitations apply when you use a line-side digital interface:

- LST1 from either of an AYC3B or an AYC11 cannot provide dial tone or stutter dial tone detection prior to dialing. This is true whether LST1 is used with or without Full CCA. (The AYC21 interface card does provide call progress tone detection for either LSE1 or LST1 and will wait for a dial tone.)

  NOTE:
  If Full CCA is installed on the system, the **tic('O')** and **tic('D')** instructions use the return codes as described in Appendix B, "Summary of Script Instructions," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761.

- Without the use of Full CCA, LST1 provided over AYC11 or AYC3B circuit cards cannot detect call progress tones (CPTs) after dialing. The AYC21 detects call progress tones when used with either intelligent or full CCA.
- When a switch is excessively loaded and a timed delay is used prior to dialing, a call can be lost if the PBX is not properly engineered and administered.
- LST1 provided over AYC11 or AYC3B circuit cards cannot detect glare conditions caused by an incoming and outgoing call attempting to use the same channel due to the lack of dial-tone detection. (Glare detection is supported by AYC21 circuit cards for either LSE1 or LST1.)
- Because the AYC3B and AYC11 circuit cards do not provide dialtone detection, the 'Dial-tone delay' parameter (see Table 3-4) must be set for a period of time that is long enough to insure that the PBX will return dialtone. Dial-tone delay may exceed a few seconds on busy PBXs. Ask the PBX administrator for the maximum delay. If you experience problems where some calls are not transferring correctly, you may need to increase the 'Dial-tone delay' parameter.

- Dial pulse is not supported on either T1 or E1 channels; however, dialing of DTMF tones is supported.
- The DEFINITY G2 does not provide forward disconnect.

Table 3-4 details the digital telephony interface specifications for line-side E1 and T1 configurations. Use Table 3-4 in conjunction with Table 3-1.

Table 3-4. Digital Telephony Interface Specifications for Line-Side Configurations

Attribute	AYC11 or AYC3B	AYC21
DS-1 framing	D4 type only	D4 for T1 & CEPT for E1
DS-1 line coding	ZCS	ZCS for T1 & HDB3 for E1
Wink-disconnect interval	300-msec default (selectable within a range of 10–2500 msec)	
Dial-tone delay	1000-msec default (selectable within a range of 20–5100 msec)	
Switch-hook-flash duration	700-msec default (selectable within a range of 10–2500 msec)	
DNIS capacity	Not supported unless used with converse vector step (CVS) or ASAI	
ANI capacity	Not supported unless used with CVS or ASAI	
Transfer capability	Flash transfers supported	

Using LSE1/LST1 for Converse Vector Step

The in-band DNIS capability is available when using the converse vector step (CVS) feature of DEFINITY on LSE1 and LST1 channels. Refer to Chapter 5, "Converse Vector Step Routing" for additional information.

Switch Integration and Administration

Switch integration for LSE1 and LST1 is done using the Digital Interfaces screen. This screen is described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. You must select Line Side Protocol for either DEFINITY or Galaxy from the Digital Interfaces screen.

In addition to the Digital Interfaces parameters, the following administration must be performed in the Analog Interfaces screen:

- In the Blind Transfer Actions field, you must set the To Initiate Transfer and To Complete Transfer fields to FP (flash and pause for a fixed delay) and H (hang-up).
- If you are using Full CCA or intelligent transfer with an AYC21 board, you must set the following under Intelligent Transfer Actions field:
  - For DEFINITY, set the To Reconnect Caller field to FPF
  - For Galaxy, set the To Reconnect Caller field to P.

Refer to Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for additional information.

Placing a card in the INSERV state allows it to be used for the purpose (play, code, etc.) for which it is allocated in the application. After performing switch integration on the E1/T1 circuit card for the LSE1/LST1 protocol, you may need to *manually* place the card into service if:

- After first installing the card or changing switch integration parameters, the voice system did not automatically place the card in the INSERV state
- The card was placed in the MANOOS state
- A diagnostic procedure failed (that is, placed that card in the MANOOS or BROKEN state).

To change the state of the E1/T1 circuit cards to INSERV, use the steps described in Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

Line-Side Connections

See "Digital Connectivity" on page 3-5 for examples of digital connection to the line side of a digital switch.

Application Development Issues

The following are LST1 and LSE1 application development issues for Script Builder, script language, and the IRAPI.

Script Builder

LST1 supports blind call origination (outcalling) and blind call transfers for DEFINITY PBXs and Galaxy ACDs only as normally performed on T/R lines. Blind transfers mean that the INTUITY CONVERSANT System will not detect call-progress tones or provide any form of answer supervision. LST1 can provide CPT detection only when used with Full CCA or when connected by means of an AYC21 board.

Because of this, when it is used with the AYC11 or AYC3B boards, LST1 does not support Script Builder requests for intelligent call originations and transfers. Furthermore, when using these boards, you must convert existing scripts which use intelligent call origination/transfer requests (DEFINITY PBXs and Galaxy ACDs only) to blind or Full CCA requests to achieve LST1 compatibility. AYC21 boards, however do support intelligent call transfer.

LSE1 does support transfers and call origination (outcalling) for calls that terminate on the DEFINITY PBXs only. LSE1 does not support call progress tone detection for calls that go outside the DEFINITY PBX.

Script Language

The following script instructions support LST1 and LSE1 operations:

- **tic('C')** (only available with the use of Full CCA or when using AYC21)
- **tic('o')**
- **tic ('O')** (only available with the use of Full CCA or when using AYC21)
- **tic('f)**
- **tic('F')**
- **tic('d')**
- **tic('D')** (only available with the use of Full CCA or when using AYC21)
- **tic('h')**

Refer to the **tic** instruction in Chapter 3, "Script Instructions," and Appendix B, "Summary of Script Instructions," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761, for additional information.

IRAPI

Refer to "T1 Application Development Issues" above for details on the supported IRAPI functions for T1 interfaces. Refer to Chapter 6 of the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761, for more information about these functions when developing IRAPI applications using LST1 or LSE1.

Primary Rate Interface

ISDN-PRI is desirable for customers that need faster call-setup times, special sigaling, or access to the information elements that are available with PRI. Such information elements as ANI, DNIS, redirecting number, and service type are available for incoming PRI calls. Outbound calls can provide information elements like outbound ANI, service type, and bearer capability.

The INTUITY CONVERSANT System supports the ISDN-PRI between itself and the digital telephone network or entity through the use of a special digital protocol, with the same physical connectivity as standard T1 digital communication. The system supports this digital ISDN communication with ISDN-PRI Layer 1 protocol rather than the T1 A/B Robbed-bit E&M Protocol used with standard T1 communications. The ISDN-PRI Layer 1 protocol uses either D4 or ESF framing. Standard T1 circuit card connectivity, as described in the previous pages, is used to implement the physical connection between the system and the remote network entity when using ISDN-PRI. PRI is also supported at the E1 rate in situations where the ACULAB protocol converter card is used.

PRI connectivity offers the ability to administer key protocol parameters through software interfaces. This parameter administration must be performed before the physical connectivity is established. Two key parameters are dependent on the framing protocol used. If D4 framing is used, line coding must be "ZCS" and D-channel inversion must be *inverted*. If ESF framing is used, line coding must be "B8ZS" and D-channel inversion must be *non-inverted*. The ISDN-PRI service provider determines the method of framing used. ESF/B8ZS is preferred.

When operating at the E1 rate, use CEPT framing and HDB3 line coding. CEPT/HDB3 are the only options allowed at the E1 rate.

The INTUITY CONVERSANT System does not support flash transfers using PRI configurations. Simulated T1 transfers can be performed only over call bridges. In both the analog Tip/Ring and digital line-side environments, the switch-hook-flash transfer releases the call from the INTUITY CONVERSANT System once the transfer is made. A call bridge, however, ties up an incoming port and an outgoing port until the call has concluded. Thus, with two ports being tied up simultaneously, more ports may be necessary.

Table 3-5 details the digital telephony interface specifications for ISDN-PRI type configurations. Use Table 3-5 in conjunction with Table 3-1 on page 3-3.

Table 3-5. Digital Telephony Interface Specifications for ISDN-PRI Type Configurations

Attribute	AYC11 or AYC3B	AYC21
DS-1 framing	D4 or ESF (selectable) for T1 rate	D4 or ESF (selectable) for T1 rate, CEPT for E1
DS-1 line coding	ZCS (with D4 framing only) B8ZS (with ESF framing only)	ZCS (with T1 D4 framing only) B8ZS (with T1 ESF framing only) HDB3 (with E1 CEPT framing only)
B-channel capacities	Up to 119 B+D when five T1 cards are used	Up to 119 B+D when five T1 cards are used Up to 90 channels when three 30B+D E1 cards are used
	(see the *Intuity* ™ CONVERSANT® System Version 6.0 *System Description*, 585-310-241, for a list of platform limitations. **These configurations are switch dependent** as not all switches support all configurations.)	
D-channel capacities	1 D-channel per system without AYC21 cards (additional D-channels are supported by AYC21 cards)	Multiple D-channels are supported up to the maximum number of T1/E1 boards. (5 channels for 5 T1 boards, 3 channels for 3 E1 boards)
Interface ID	1 (for a card with a D-channel, not selectable) 2-5 (for a card without a D-channel)	
DNIS capacity	0–15 digits	
ANI capacity	0–15 digits	
D-channel backup	Not supported	
Transfer capability	Not supported	

PRI Provisioning

Supported B-channel capacities in PRI configurations are switch dependent (see Table 3-5) as not all switches support all configurations. For example, the 5ESS switch only supports the 23 B+D configuration, but the 4ESS switch can support up to 119 B+D. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *System Description*, 585-310-241, for information on supported PRI configurations.

Special parameter provisioning of PRI is required on the switch, but is not part of the normal order process for AT&T PRI network services. Thus, give special attention to the determination and provisioning of these parameters when ordering and implementing this feature. In addition, the INTUITY CONVERSANT System uses some Layer 2 and Layer 3 parameters that must be correct and matching in both machines. Table 3-6 and Table 3-7 show how to set these parameters on the switch.

You should provision incoming calls to the CONVERSANT System so that the channel number is exclusive and not preferred. Also, if the switch is configured to deliver automatic number identification (ANI) on a subscription basis, it is not possible for the system to request a different type of ANI on a call-by-call basis.

Table 3-6. PRI Layer 2 Parameters

Parameter	Value
Retry Count N200	3
Timer T200	1 sec
Timer T203	30 sec
High-level data link control or HDLC (D4/ZCS)	Inverted
HDLC (ESF/B8ZS)	Noninverted

Table 3-7. PRI Layer 3 Parameters

Parameter	Value
Timer T302*	15 sec
Timer T303*	4 sec
Timer T304*	30 sec
Timer T305*	4 sec
Timer T308*	4 sec
Timer T309*	30 sec
Timer T310*	10 sec
Timer T313*	4 sec
Timer T316*	120 sec
Timer T3M1*	120 sec
Interface ID (with D-channel)	1
Interface ID (without D-channel)	2 - 5
Bearer capability	64 Kbps voice

* All timers are adjustable as described in the /vs/man/cat4/pri.rc.4 file.

PRI Switch Integration and Administration

Switch integration for the PRI feature is done using the Digital Interfaces screen. This screen is described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. You must select ISDN-PRI Layer 1 Protocol from the Digital Interfaces screen.

(Unless your digital circuit cards are AYC21s, you must assign one SP circuit card to process the PRI protocol besides the T1 circuit cards required to provide the physical layer for the PRI.)

To assign PRI functionality to an SP circuit card, refer to Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.To assign PRI functionality to a T1 or E1 circuit card, refer to Chapter 6, “Switch Interface Administration,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

PRI Connections

As mentioned earlier PRI can be connected through either of a T1 or E1 circuit card. See "Digital Connectivity" on page 3-5 for examples.

Understanding B-Channel and D-Channel

The D-channel cannot be used to run applications. It carries messages between the switch and the system. These messages are used to control the state of calls on all the other PRI channels.

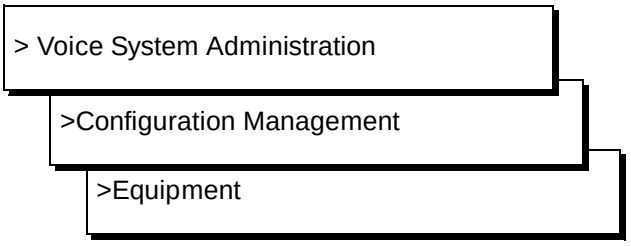
All the other PRI channels are referred to as B (bearer) channels. The B-channels provide two-way audio channels to run applications. Therefore, on a PRI that has been configured to have only one T1 circuit card, the first 23 channels (B-channels) on that card can be used to run applications. The 24th channel (D-channel) is reserved for call control. If your PRI is configured with more than one T1 card, the additional T1 cards (the ones configured without a D-channel) will have 24 B-channels on which to run applications. The system can run applications on a total of 119 B-channels (that is, five T1 cards). To provide acceptable performance, only 96 B-channels can be used for incoming calls; the rest of the channels must be used for outgoing bridged calls.

For E1 PRI, channel 0 is reserved for framing and channel 16 is reserved for the D channel, the remaining 30 channels are B channels. Typically, each E1 PRI interface has its own D channel (unlike T1 PRI where a single D channel frequently controls more than one T1 interface).

Determining the D-Channel

If you do not know which channels have the D-channels, perform the following procedure. Refer to Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for more information.

1. Begin at the CONVERSANT Administration menu, and select the following sequence:



   The Voice Equipment screen displays a list of all channels in the system.

2. Use the ▲ and ▼ cursor keys to scroll through the list of channels.

   The D-channels are the only channels that are labeled PRID in the “TYPE” column. B-channels are labeled PRIB.

Once you know which channels have the D-channel, you are ready to bring the PRI into service to allow it to begin taking calls. Change the state of all PRI channels to INSERV using the steps described in Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

Display the Options field to see with which D-channel group the card is associated (PRI1 PRI2, etc.) and whether it has a D-channel (DCHAN).

PRI Application Development Issues

The following are PRI application development issues for Script Builder, script language, and the IRAPI.

Script Builder

The PRI feature provides the following Script Builder external actions and an external function for use in PRI applications:

- The ISDN_billing external action provides the billing number to incoming call applications.
- The Attr_ANI external function allows an application to request the billing number for incoming calls on a call-by-call basis.

  NOTE:
  The Attr_ANI external function is not necessary for facilities that subscribe to ANI.

- The ISDN_service external action allows an application to choose Service Type for outgoing PRI calls.

In addition, PRI supports the following call-control Script Builder actions:

- Answer
- Disconnect
- Make Call
- Call Bridge

The Call Transfer action is not supported for PRI because the PRI protocol does not support the transfer function.

For additional information about integrating the PRI feature in your CONVERSANT application, refer to Chapter 6, "Using Optional Features with Script Builder," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760.

Script Language

Several capabilities are available to implement the PRI feature in TSM script language applications.

- The **tic** instruction is used for basic control of incoming and outgoing calls on the PRI. The **tic('C')** and **tic('O')** instructions provide additional return code information over the T1 and analog interface implementations.

  The following additional script registers apply to PRI:

  - IE.ANI – Calling party number
  - IE.DNIS – Called party number
  - IE.REDIRECTING – Originally dialed number
  - IE.SERVICE – Incoming service type

- The **setattr** instruction can be used to request the Calling Party Number (CPN) from the network before starting the script.
- The **setstring** instruction can be used to send a CPN on an outbound call.
- The **setparam** instruction can be used to specify an outbound service type or bearer capability on an outbound call.

For additional information about integrating the PRI feature using TSM script language, refer to Chapter 3, "Script Instructions," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761.

INTUITY Response API

The **irCall()**, **irAnswer()**, **irDial()**, and **irDisconnect()** functions provide the basic call control capabilities for T1 interfaces. The **irFlash()** and **irStartSpeechED()** function is not supported for PRI interfaces. The **irSetIE()** and **irGetIE()** can be used to set and get information elements available only with PRI. Refer to Chapter 6 of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761 for more information about these functions when developing IRAPI applications.

Advanced PRI Capabilities

For additional information about PRI and for sample applications of advanced PRI programming, refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Advanced PRI Developer's Guide,* comcode 107911497. This document is restricted to Independant Software Vendors (ISVs) and Value Added Resellers (VARs) and they can use it to create packages that provide additional PRI capabilities for their customers: the end users.

This document is intended for the INTUITY CONVERSANT application developer who wishes to develop applications that go beyond the standard Commercial PRI capabilities that are documented above and elsewhere. It describes how to extend the capabilities of the standard Commercial PRI services and how to develop General Purpose PRI application processes that offer even more complete control of the ISDN PRI signaling.

Adjunct/Switch Application Interface

4

What's in This Chapter

This chapter describes the use of the INTUITY CONVERSANT Voice Information System Adjunct/Switch Application Interface (ASAI) feature and the requirements that must be met to implement this interface. Also provided are ASAI application and call flow examples, a discussion of the use of ASAI versus the DEFINITY converse vector step (CVS), and a list of application development issues that must be addressed when using ASAI.

ASAI Overview

Briefly, ASAI provides an Integrated Services Digital Network (ISDN)-based interface between switches and adjunct processors. The CONVERSANT System's ASAI feature supports this application interface for communications with the Lucent DEFINITY Communications System, Generic 3 (hereafter referred to as DEFINITY G3). This digital signaling interface allows the CONVERSANT System to monitor and route calls on the DEFINITY G3. When used in conjunction with Tip/Ring (T/R) or digital Line Side E1 or T1 interfaces (LSE1/LST1), the ASAI interface allows the system to monitor and control the incoming calls it receives.

Advantages of Using the ASAI Feature

When using ASAI, 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 CONVERSANT T/R or LST1/LSE1 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 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 ability reduces both call holding time and customer frustration. This benefit holds true even when calls are transferred several times or are transferred between live agents.

The 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 CONVERSANT piece of the overall CONVERSANT/host application. Such ASAI information as ANI and DNIS related to a particular call can be retrieved for use in the script handling the call. Refer to "ASAI Application Development" on page 4-10 for more information.

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 Connectivity

An ASAI link between the DEFINITY G3 and the CONVERSANT System, delivers control and supervisory messages about each T/R, LSE1, or LST1 channel. The

link may be realized either as an ethernet connection or as an ISDN-BRI data-channel connection. One ASAI link per CONVERSANT System is supported.

Generally, such a configuration looks like .Figure 4-1

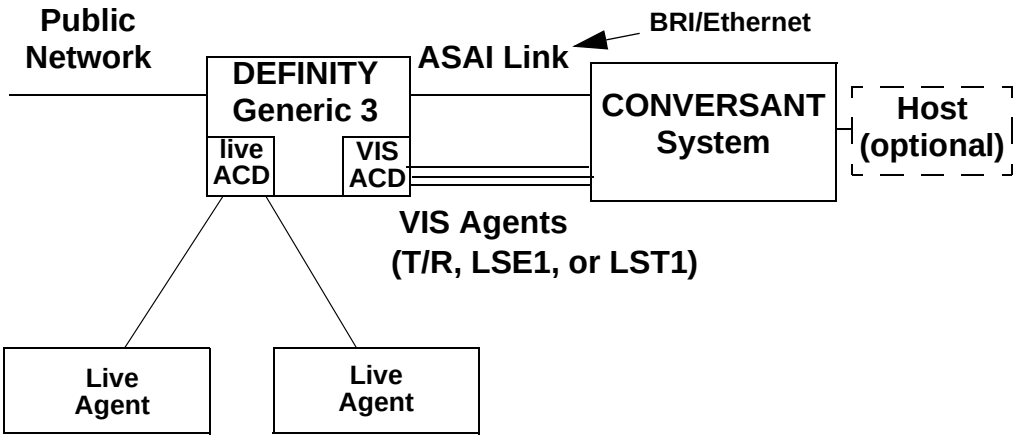


Figure 4-1. Typical CONVERSANT System and DEFINITY G3 Configuration

NOTE:
The public network must provide an ISDN-PRI connection to the DEFINITY G3 for an application to receive calling number information.

Establishing an Ethernet ASAI Link

In cases where the ASAI link is realized as an ethernet connection, an ethernet expansion card in the CONVERSANT System is connected to a DEFINITY LAN Gateway circuit pack in the DEFINITY G3.

(For more information about the LAN Gateway, refer to: *DEFINITY Communications System Generic 3 Installation, Administration, and Maintenance of CallVisor ASAI over the DEFINITY LAN Gateway,* Issue 1, 555-230-223.)

Figure 4-2 shows a typical LAN configuration.

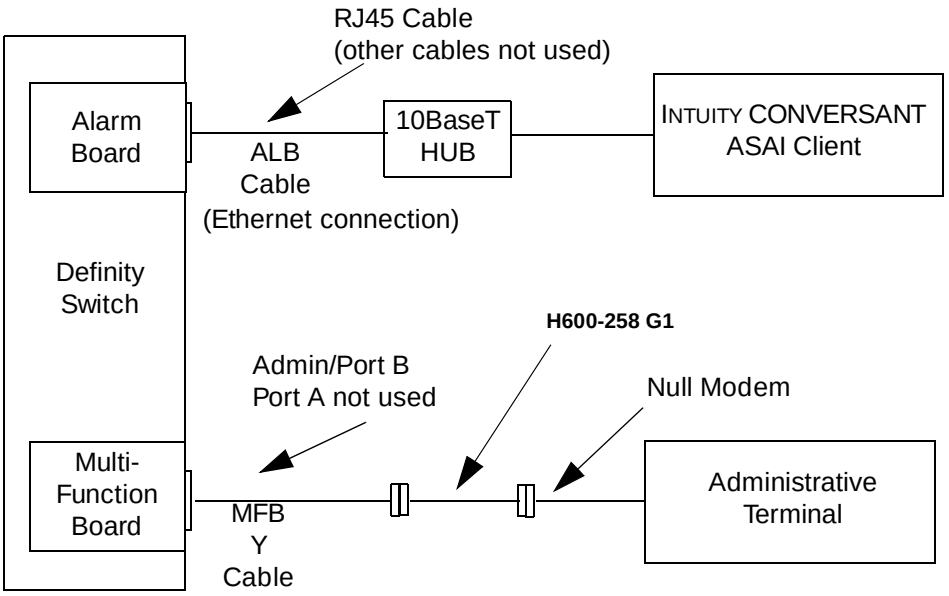


Figure 4-2. Typical LAN Wiring for an ASAI Link

For information about connectivity to DEFINITY, refer to: *DEFINITY Communications System Generic 3 CallVisor ASAI Planning Guide,* Issue 4, 555-230-222.

Establishing a BRI D-Channel ASAI Link

The ASAI D-channel between the TN556 ISDN/BRI circuit card in the PBX and the IPCI circuit card in the CONVERSANT System uses a separate connection. Figure 4-3 shows the D-channel connectivity.

You must connect a Lucent 440A4 8-pin terminating resistor (or equivalent) to the LINE connector of the IPCI circuit card using the D8W cable provided. Use another D8W cable to connect from the connecting block to the terminating resistor.

NOTE:
Total cable length from the DEFINITY G3 System to the CONVERSANT System must not exceed 580 meters (1900 feet). Refer to "ASAI Administration" on page 4-6 for additional information about administration of this communication arrangement.

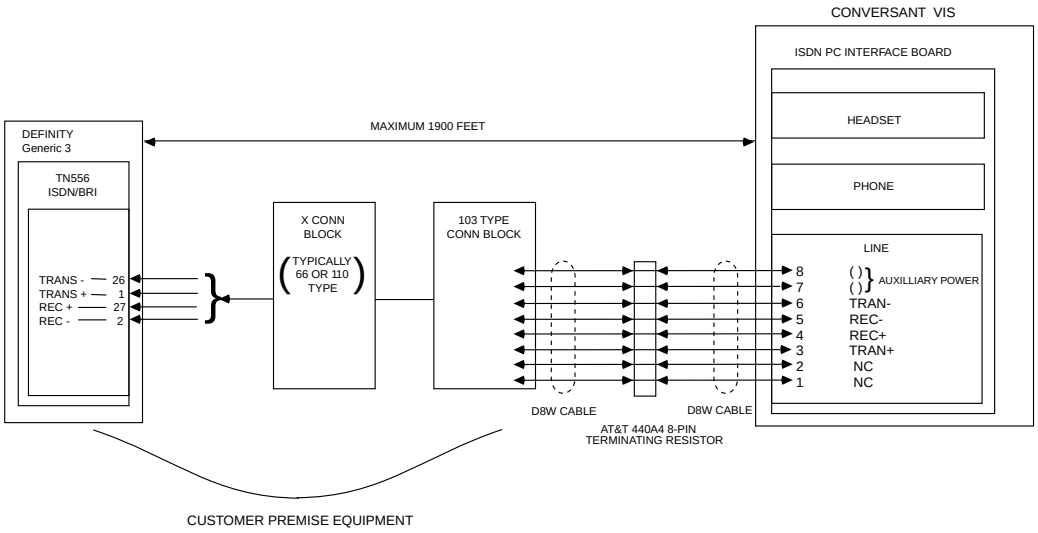


Figure 4-3. Typical D-Channel Wiring for an ASAI Link

Connecting the CONVERSANT System Agents

The following information details making T/R and LST1 connections from the CONVERSANT System to the PBX.

Analog T/R Connections

ASAI can be provisioned using analog T/R lines between the PBX and the CONVERSANT System. Analog T/R circuit cards must be installed in the CONVERSANT System and each line connected separately. Refer to the ***Intuity*** ™ CONVERSANT® System Version 6.0 ***System Description***, 585-310-241, for information on T/R circuit card capabilities for ASAI.

Line-Side Digital Connections

ASAI can also be provisioned with line-side T1 or E1, which allows digital connections between the CONVERSANT System and the line side of the PBX. This type of connection allows the utilization of various PBX features, such as call transfer and call progress tone (CPT) detection (either in conjunction with Full CCA or where an AYC21 interface board is used for communications), which are not compatible with an ordinary T1 trunk connected between the CONVERSANT System and PBX.

Analog configurations require 24 separate connections to support an identical configuration provided by one LST1 cable. It requires 30 analog connections to compare to one LSE1 connection. There is also a significant reduction in the number of circuit cards required to support the interface: one E1 circuit card supports the same amount of traffic as five IVP6 circuit cards.

ASAI Administration

Administering the ASAI feature is a four-step process. The following example assumes you are installing a voice response application with a configuration in which calls placed to an Automatic Call Distributor (ACD) on the PBX are directed to (agent) lines on the CONVERSANT System. The CONVERSANT System is used to select a service for the incoming call based on the dialed number identification service (DNIS), or called number. The service requests the DNIS number and automatic number identification (ANI), or calling number) from the ASAI interface and uses this information as part of the service being provided to the caller. To administer the ASAI feature, perform the following steps on the switch and the CONVERSANT System:

1. Install and administer the BRI or Ethernet board. Refer to the hardware installation book for your platform for additional information. (Station administration is the same for either. See Table 4-1 on page 4-7.)
2. Administer the ACD domain (hunt group) on the CONVERSANT System and the DEFINITY G3.
3. Administer the T/R, LSE1 or LST1 telephone lines.
4. Administer the CONVERSANT System agent lines.

Once you have completed these steps, you can assign services to DNIS numbers. Refer to Chapter 3, “Configuration Management” of the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for information on how to assign these services.

NOTE:
The following procedures assume you have installed the necessary hardware on the CONVERSANT System and the DEFINITY G3. Refer to “ASAI Connectivity” earlier in this chapter and the hardware installation book for your platform for additional information.

NOTE:
The following procedures assume that you have completed the necessary administration on the PBX. Refer to the *DEFINITY Communications System Generic 3i Implementation*, 555-230-650, for additional information.

BRI/Ethernet Administration

With either a new CONVERSANT System installation or an upgrade, you must administer the DEFINITY ACD split to be used for ASAI connectivity between the DEFINITY and the CONVERSANT System. Administration is the same for either of a BRI or Ethernet connection. Use the DEFINITY **add station** or **change station** commands to administer the ACD split. Use Table 4-1 for appropriate values.

Table 4-1. Administration Field Name and Requirements

Field Name	Required or Optional?	Contents
Extension:	Required	Whatever fits your dial plan
Type:*	Required	ASAI
Port:	Required	The port that connects to the ASAI line
Name:	Optional	Can be used as an identifier
XID:*	Required	y
Fixed TEI:*	Required	y
TEI:*	Required	3
MIM Support:*	Required	n
CRV Length: *†	Required	2

*To match the built-in administration of the IPCI or Ethernet circuit cards and the ASAI software, the Type, XID, Fixed TEI, TEI, MIM Support, and CRV Length fields *must* have the contents indicated above.

†In some previous releases, the CRV Length field required a value of 1. You *must* use the value 2 for INTUITY CONVERSANT *System* V6.0.

Administering the ACD Split Domain

On the CONVERSANT System

You must administer the ASAI feature to monitor the ACD hunt group extension and allow the CONVERSANT System to receive information on calls placed to its CONVERSANT System agent lines.In other words, you must administer the ASAI feature on the CONVERSANT System 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 CONVERSANT System agent lines. This domain is referred to as the *CONVERSANT System ACD domain*. You can administer only one CONVERSANT System ACD split domain on the system. Therefore, all

CONVERSANT System agent lines must be part of a single ACD split. Figure 4-1 on page 4-3 shows this configuration. Refer to Chapter 4, “Feature Packages,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, to administer the CONVERSANT System ACD Split Domain.

On the DEFINITY

Use the DEFINITY **add hunt group** or **change hunt group** command to administer the BRI or Ethernet line. Table 4-1 lists the values required for proper implementation of the DEFINITY for the ASAI link. Table 4-2 Shows a typical way of administering a DEFINITY hunt group to be an ACD. Use the following DEFINITY call center documentation to provide administration details: *DEFINITY Communications System Generic 3 Call Vectoring/Expert Agent Selection (EAS) Guide*, 555-230-520.

Table 4-2. DEFINITY Hunt Group Field Name and Values

Field Name	Contents: Non-EAS	Contents: EAS
Group Number:	The number of the hunt group	
Group Extension:	The extension to be used as the lead for the hunt group	
Group Type:	ucd	
ACD?	y	
AAS?	n	
Vector?	n	y
Controlling Adjunct:	none	

Administering the T/R, LSE1, and LST1 Lines

Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, describes how to administer T/R, LSE1, and LST1. To be certain that you select options that are compatible with the DEFINITY G3 (only certain versions) System, select the “DEFINITY” item in the PBX Defaults screen. DEFINITY 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 Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. These lines or channels are referred to in the following text as *CONVERSANT System Agent lines*.

CAUTION:
Do not proceed until the lines have been placed in the inserv state.

Administering the CONVERSANT System Agent Lines

After creating and bringing the ACD split or VDN domain into service and administering the T/R and LST1/LSE1 lines, you must administer and log in a CONVERSANT System in one of two ways: as an agent extension in an ACD split, or by an agent ID (with optional password) in an EAS environment. This is required if your service is going to use DNIS or the **A_Callinfo** or the **A_tran** actions described in Chapter 8, "Using Optional Features," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760. If you do not log in an agent line, 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 CONVERSANT System by the ACD.) Refer to Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for how to log in the CONVERSANT System agent lines. Refer to Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, to assign DNIS service to channels.

DEFINITY System Planning

DEFINITY System planning involves defining what changes you must make to the DEFINITY software configuration and ACD environment to support the planned applications. The following is a list of items to consider when planning for the changes.

- Call vectoring is strongly recommended for use in implementing all CONVERSANT System ASAI applications. This is especially true for data screen delivery applications that involve agent-to-agent transfers or DNIS service and for voice response applications that make use of DNIS service.
- Call vectoring is mandatory for routing applications. Call vectoring is also mandatory for data screen delivery applications that 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 want to aggregate agents currently in multiple splits into a single split. This minimizes the number of domains that the CONVERSANT System monitors 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.

ASAI Application Development

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.

NOTE:
The CONVERSANT System 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 CONVERSANT System ASAI feature to log T/R, LSE1, or LST1 channels in and out of an ACD split on the DEFINITY G3.

The following application development issues must be considered when implementing the ASAI feature with the CONVERSANT System:

- Types of ASAI applications
- Using ASAI versus the Converse Vector Step (CVS)
- Using ASAI in a Call Center
- CONVERSANT System script design
- Call-flow design
- Host-application design

ASAI Application Types

The capabilities provided by the ASAI feature support three classes of applications:

- Voice response applications
- Routing applications
- Data screen delivery applications

These classes of applications can all run simultaneously on a CONVERSANT System. This implies that an *INTUITY* CONVERSANT ASAI System provides coresident voice response and DEFINITY G3-to-host gateway capabilities. A single call, for instance, can first be routed by the CONVERSANT System, handled with a voice response application on the CONVERSANT System, and then be monitored by the same s;ystem as the call is ultimately delivered to a live agent. Furthermore, integration of the voice response and gateway capabilities allows agents to interact with callers based on the data collected in a voice response script through a host screen. The delivery of a data screen to an operator that contains information about the incoming caller is called a “screen pop.”

ASAI Voice Response Applications

In voice response applications using the ASAI feature, incoming calls can be routed to the CONVERSANT System over T/R, LSE1, or LST1 channels via an ACD split on the DEFINITY G3. Figure 4-4 shows this class of application.

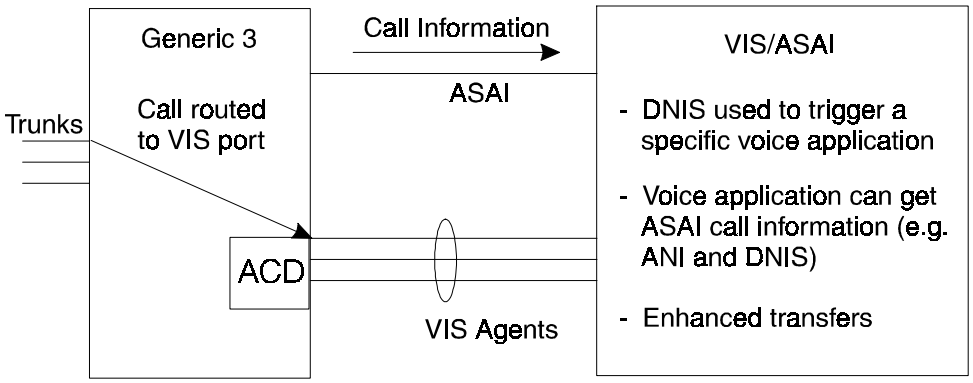


Figure 4-4. ASAI Voice Response Applications

As a call is delivered to the CONVERSANT System, it receives ASAI information related to the call through the D-channel connected to an IPCI circuit card in the INTUITY CONVERSANT System. ASAI allows it to receive the DNIS and/or ANI information of an incoming call to an analog T/R or digital LSE1 or LST1 line over this D-channel. The DNIS and ANI information can be used to control the voice application used for the call. The ASAI information related to the call is made available to the specific voice application that interacts with the caller. In addition, the call control capabilities of ASAI can be used to transfer the call away from the CONVERSANT System if the caller needs to speak to a live agent. The ASAI feature provides the following for voice response applications:

- Channel sharing — The DNIS and/or ANI information associated with the incoming call is used to select a particular Script Builder script to service the call. This allows T/R, LSE1, and LST1 ports to be shared across many applications. With port sharing, the same number of ports 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.
- ANI service — Providing this service allows scripts to be customized according to the calling party number or a range of numbers (for example, an area code).

- Call information — Once the call has been answered by the CONVERSANT System, 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 G3 ASAI feature of direct agent calling can be used to transfer the call. 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 data screens to agents that are based on data collected by the voice script that previously serviced the caller and any combination of ANI and/or DNIS information. Refer to "Data Screen Delivery Applications" later in this chapter. The availability of ANI for script selection or within the voice script permits the design of unique voice response applications. Examples include:
  - Locator service. A local or host database can be used to determine the closest car dealers, ATMs, stores, etc.
  - Weather reports. A weather report for the caller's area can be provided.
  - Pay-per-view. A cable company can use ANI to automate customer selection and billing of pay-per-view programs.
  - Caller-dependent transfers. The full 10-digit ANI can be used to identify callers and determine where they should be transferred if they need to speak to a live agent. This is 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 using the ASAI feature, the CONVERSANT System is used as a routing server to support the routing capabilities of ASAI and the call-vectoring feature on the DEFINITY G3. Figure 4-5 shows how a routing application on the CONVERSANT System receives and responds to call-routing requests sent by the DEFINITY G3. The application uses routing information provided by the CONVERSANT System to direct the call to a live agent or to a CONVERSANT System agent via either a T/R, LSE1, or LST1 connection.

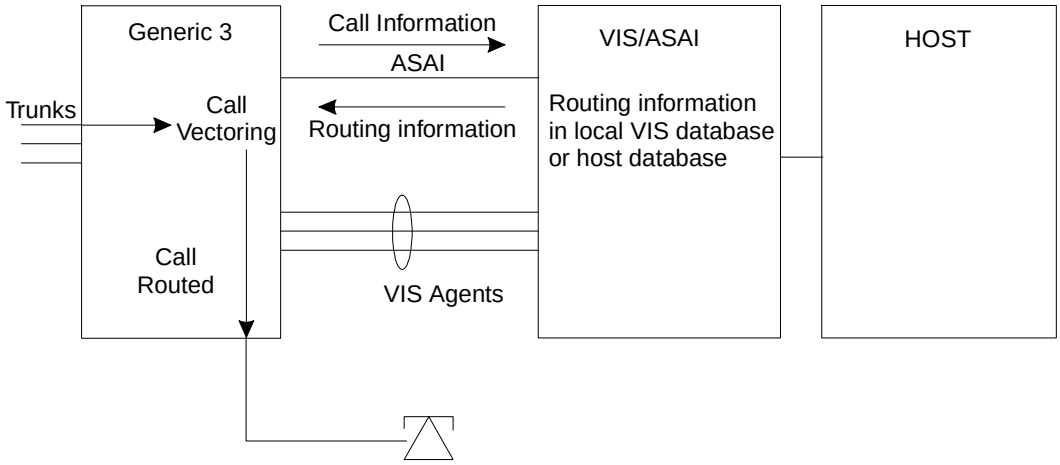


Figure 4-5. ASAI Routing Applications

These call-routing requests are generated by the DEFINITY G3 when a call is processed by specific call vectors on the DEFINITY G3.

Information as to where to route calls can reside on the CONVERSANT System in a local database or can be provided by a host to which the CONVERSANT System is connected. Call-routing is typically based on ANI or call-prompting data collected by the DEFINITY G3.

The use of routing capabilities can significantly improve the efficiency of a call center as shown in the following examples.

- Priority service — Important or “priority” callers such as major clients can be routed to a common agent group but queued at a higher priority so that they are serviced faster. These callers can also be routed to the specific agent who 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 before being connected to an agent so that no network costs are incurred.
- Geographically-based service — 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, an application that resides on the host delivers a specified data screen related 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. Figure 4-6 shows a data screen delivery application.

NOTE:
Data screen delivery applications are also known as *coordinated voice/data screen delivery* or *screen pop* applications.

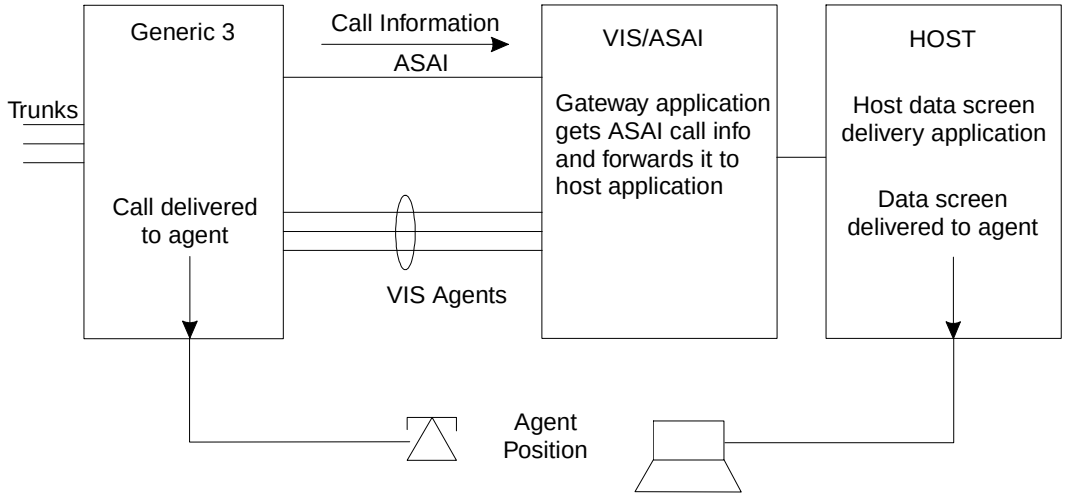


Figure 4-6. Data Screen Delivery Applications

Note that the delivery of data screens is not a function of the CONVERSANT System itself. The system acts only as a communications gateway between the DEFINITY G3 and the host computer. A monitoring application on the CONVERSANT System provides the ability to track the status of calls on the DEFINITY G3. This monitoring application receives information about calls delivered to live agents and forwards this information to the application on the host. 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 G3 call-prompting information collected from the caller. In addition, the call may have been serviced by a CONVERSANT System voice script and then transferred to a live agent. In this case, information collected in the voice script can be saved and passed to the host at the time the call is delivered to the agent.

Monitoring applications on the CONVERSANT System can therefore be used to support data screen delivery for three different call-flow scenarios:

- CONVERSANT System-to-agent transfers — In this scenario, calls are delivered to the system and then transferred to a live agent. As described previously, 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 and DNIS and call-prompting information collected by the DEFINITY G3.
- Incoming call directly to agent — In this scenario, incoming trunk calls are delivered directly to live agents. Data screens delivered to agents are based primarily on ANI and DNIS and/or call-prompting information. Data screens are not based on data collected in a voice script, since a voice script is not used to collect data from the caller.
- Agent-to-agent transfers — In this scenario, calls are transferred between live agents. Here, for example, "screening" agents can be used to collect information from the caller and handle simple transactions. The call can subsequently be transferred to "specialized" agents to 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 specialized agents who can 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 also available in this scenario. This information can be used in conjunction with data entered by a live agent to provide the basis for data screens.

  NOTE:
  You must plan your call flows carefully if you are using multiple ASAI adjuncts with the same DEFINITY G3 System. Once a call is monitored by a particular CONVERSANT System, the call cannot be redirected or transferred to a domain monitored by another system 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 CONVERSANT System that monitors calls delivered to them. Also, for example, you may have CONVERSANT System-to-agent transfers to support data screen delivery based on CONVERSANT System collected data. In this case, you should configure multiple CONVERSANT Systems to "front end" mutually exclusive sets of live agents. These considerations do not apply if you are using only one CONVERSANT ASAI System and it is the only ASAI adjunct.

The CONVERSANT System-to-agent transfer scenario described above is supported 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.

The ability to save voice script data is useful in many ways. 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 useful than ASAI information such as ANI to both the host application and the live agents handling calls.

The ability to save voice script data with the enhanced transfer capability provides a useful bridge between voice-response and data-screen delivery applications. It provides true integration (in addition to coresidency) of the voice-response and PBX-to-host gateway capabilities offered with the CONVERSANT System's 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 that is delivered to an agent). With the 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.

ASAI Versus Converse Vector Step

The Converse Vector Step (CVS) allows the PBX to maintain control of a call while capabilities of the CONVERSANT System are being used. Whether to use ASAI or the CVS depends on several factors, including cost, traffic, and desired functionality. For example, the CVS feature, used in a script, could support a low cost ANI routing application. Large traffic volumes may require an ASAI-based solution due to the more efficient ASAI adjunct routing. Refer to Chapter 5, "Converse Vector Step Routing" for additional information about the CVS.

The following provides a list of the capabilities and limitations of using the two features on T/R, LSE1, or LST1 lines.

- Both ASAI and CVS provide the delivery of ANI, DNIS, and switch call prompting digits for T/R, LSE1, or LST1 calls. The CVS provides this information on an in-band basis while ASAI makes the data available on an out-of-band basis. The ASAI out-of-band exchange of data is faster.

  NOTE:
  CVS allows a maximum of two parameters to be delivered.

- The CONVERSANT System's ASAI actions **A_Event** and **A_RouteSel** can be used in monitoring and routing scripts even if the calls are delivered via the CVS.

In addition, both ASAI and CVS have some unique properties that may influence the decision as to which feature to use:

- ASAI properties
  - — When the CONVERSANT System is used as a gateway for PBX-to-host applications, the **A_Tran** action simplifies call-flow development using screen pops based on CONVERSANT System collected data.
  - — Dynamic port allocation is simpler because ANI and DNIS service administration is supported. (Some script programming is necessary if you are using CVS for port allocation. For example, you could write an INTUITY Response Application Programming Interface (IRAPI)-based start-up script written to obtain ANI and DNIS for the CVS interface and then "exec" the appropriate script for that ANI/DNIS information; however, that IRAPI application is not provided with the generic software.
- CVS properties
  - — CVS allows a call to remain in a live agent queue while interacting with the CONVERSANT System.
  - — Queue position and administered digit string can be passed to the CONVERSANT System using CVS. Queue position could be used as the basis for an anticipated delay announcement. An administered digit string could be used to identify specific annoucements to be played to callers.

Using ASAI in a Call Center

ASAI can significantly improve the operations in a call center. This feature provides the following benefits:

- Enhanced customer service

  Caller- 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 ensure 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.

  In data screen delivery applications, information associated with a given call is available to each agent receiving the call. This eliminates the need for callers to repeat information to each agent. For example, a caller may be directed initially to a CONVERSANT System T/R, LSE1, 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

CONVERSANT System's ASAI feature, the identity of the caller and additional information collected from the caller by the voice-response application can be saved and presented in a data screen to the live agent receiving the transferred call. This eliminates the need for the caller to repeat information already collected when calls are transferred multiple times or are transferred between live agents. Thus, call-holding time is reduced.

- Improved price/performance

  The coresidency of voice-response and PBX-to-host gateway applications with the 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 CONVERSANT System's piece of the overall CONVERSANT System/host application. In addition, the use of DNIS in voice-response applications to enable T/R, LSE1, or LST1 channel sharing means that the same number of CONVERSANT System channels can service more calls.

- Reduced cost of doing business

  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 use of data-screen-delivery applications reduces the time needed to service calls. Because calls are shorter, 800 network charges are lower. 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).

Specific agent tasks may change when you add an ASAI application such as data screen delivery 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. Be sure to explain the benefits of the application so that agents can take advantage of them.

If data screen delivery is performed for agent-to-agent transfers, carefully read the information on "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. 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 is 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”). The agent must be sure to place the original call on transfer hold (not regular hold) before completing the transfer. If the agent used regular hold, the agent would be unable to return to the original caller.

Use the following procedure 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 ensures that the CONVERSANT System 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 original caller 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. Agent 1 dials Agent 2 while hearing dial tone on the new, active call appearance.
3. 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.
4. 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.
5. 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. Agent 1 goes to the next step and ignores the dial tone).

6. Agent 1 makes 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.
7. Agent 1 completes the transfer by hitting the Transfer button again. This is the third time Agent 1 has pressed the Transfer button.

CONVERSANT System Script Design

The CONVERSANT System ASAI feature provides four additional Script Builder actions that are used to access ASAI capabilities. These actions are discussed in detail in Chapter 8, "Using Optional Features," in *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760. A brief summary of these actions is provided below:

- **A_Callinfo** — This action is used within a voice response script to retrieve ASAI information about a call delivered to a T/R, LSE1, 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 CONVERSANT System.
- **A_Event** — This action is used within routing scripts to receive information about call-routing requests sent by the DEFINITY G3 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.
- **A_RouteSel** — This action is 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 CONVERSANT System to send ASAI call routing information to the switch.
- **A_Tran** — This action is used within a voice-response script to transfer a call away from a T/R, LSE1, or LST1 channel on the CONVERSANT System. 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_Tran** actions 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. "ASAI Application Examples" below includes an example of a voice script making use of the **A_Callinfo** and **A_Tran** events.

The **A_Callinfo** and **A_Tran** actions are used only in voice scripts that handle calls delivered to a CONVERSANT System T/R, LSE1, 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 T/R, LSE1, or LST1 channel.

For ASAI voice response applications, incoming calls are routed to the CONVERSANT System over T/R, LSE1, or LST1 channels configured either as extensions in an ACD split or as agent ID's under a Vector Directory Number (VDN) in an EAS environment on the DEFINITY G3 System. The CONVERSANT System uses the Notification capability of ASAI to monitor the ACD split or VDN. As a call is offered, the CONVERSANT System receives event reports indicating the status of the call (for example, call offered, queued, alerting, and connected event reports). The CONVERSANT System uses the information contained in these event reports to provide the following capabilities:

- DNIS and ANI service — The DNIS and/or ANI 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 is typically represented by a unique Vector Directory Number (VDN) on the DEFINITY G3 switch. Calls to these different VDN's can be routed to the same CONVERSANT System split. The DNIS and/or ANI information associated with an incoming call is then used to select a particular application. An administrative screen on the CONVERSANT System allows the different dialed numbers to be associated with a specific voice response application. This allows T/R, LSE1, 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 CONVERSANT System answers the call, 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 obtain 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 CONVERSANT System. 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 and/or ANI 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, LSE1, 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, LSE1, 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 switchhook 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 G3 direct agent calling feature can only be invoked via ASAI.
- Information captured in the voice script can be saved for subsequent use in a data screen delivery application. Information assigned to the CONVERSANT System Data field of **A_Tran** is saved by the CONVERSANT System even after the voice script terminates. The CONVERSANT System associates this data with the transferred call and makes this data available in call events passed to the monitoring script that monitors the transferred call.

The third enhancement is very useful for data screen delivery applications where the screens delivered to agents are based on data collected by the CONVERSANT System. 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 CONVERSANT System 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 G3 System. In routing scenarios, calls are not physically delivered to T/R, LSE1, or LST1 channels on the CONVERSANT System. Instead, incoming calls to the DEFINITY G3 are directed to a vector containing an *adjunct route* step. The adjunct route step causes a *route request* message to be sent to the CONVERSANT System. The route request message contains information pertaining to the call (for example, ANI). The CONVERSANT System uses this information to determine where to route the call.

After the CONVERSANT System determines where to route the call, a *route select* message is sent back to the DEFINITY G3 System. The route select message contains a destination address provided by the CONVERSANT System that the DEFINITY G3 uses to further direct the call. In routing scenarios, the

CONVERSANT System may be viewed as a routing server which the DEFINITY G3 calls upon to route calls processed with a routing vector.

Note that as a result of routing, the call may be directed to a CONVERSANT System T/R, LSE1, 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 is not sufficient to identify the caller (for example, ANI not recognized).

Routing applications on the CONVERSANT System are supported through the use of routing scripts that 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 G3 System up to the script level. The **A_Event** action returns a ROUTE REQUEST event when the DEFINITY G3 System sends such a message. If no route-request messages are sent, the **A_Event** action waits until it receives one. When a ROUTE REQUEST event is made available to the script, it reflects information in an ASAI route-request message sent by the DEFINITY G3 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 G3 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. These events can arrive at any point in time (controlled by vector processing on the DEFINITY G3 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 CONVERSANT System. Once the **A_Event** action retrieves a ROUTE REQUEST event, subsequent actions below are executed.
2. Other standard Script Builder actions that make use of the data made available in the ROUTE REQUEST event to determine where to route the call. 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 CONVERSANT System. This causes an ASAI route select message containing the routing information to be sent to the DEFINITY G3 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 below in “ASAI Application Examples.”

A routing script may not contain any Script Builder actions that pertain to voice response capabilities (Announce, Prompt and Collect, etc.). A routing script is assigned by using the "RTE" domain designation as described in Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

A routing script can use any of the information returned in the ROUTE REQUEST event. To route the call, refer Chapter 8, "Using Optional Features," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760. 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 [DDD] number). Also, the call may be routed to a specific agent within an ACD split (direct agent routing). To do this, set the Destination Number field in **A_RouteSel** to the desired agent extension and 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 CONVERSANT System 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 CONVERSANT System 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 CONVERSANT System itself.

In data screen delivery applications, calls are not physically delivered to a T/R, LSE1, or LST1 channel on the CONVERSANT System. Rather, calls are delivered to ACD agents on the DEFINITY G3 System. Note, however, that a call may have previously been delivered to a T/R, LSE1, or LST1 channel to collect information from the caller.

Events

Use the **A_Event** action to design a monitoring script. When used in monitoring scripts, the **A_Event** action returns the following types of call events:

- CONNECT Event — This event indicates that a monitored call is being delivered to an agent.

- ABANDON Event — This 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 — This 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 8, "Using Optional Features," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760. The three call event types passed to a monitoring script reflect information contained in ASAI event reports for the call.

General Structure

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 that can arrive at any point in time as determined by how and when calls progress on the DEFINITY G3 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 CONVERSANT System. 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 below in "ASAI Application Examples."

A monitoring script may not contain any Script Builder actions that pertain to voice-response capabilities (Announce, Prompt and Collect, etc.). To assign a monitoring script, use the "VDN", "ACD", or "CTL" domain designation as described in Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

A monitoring script can 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 can 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).

If you make changes to an existing monitoring script or add a new monitoring script, you must do one of the following:

1. Disable and then re-enable the domain. Refer to Chapter 4, “Feature Packages,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.
2. Stop and restart the voice system to activate the new script. (Refer to the appropriate maintenance book.)

Call-Flow Scenarios

Monitoring scripts on the CONVERSANT System can be used to support data screen delivery for the following three different call-flow scenarios:

- CONVERSANT System-to-agent transfers — In this scenario, calls are initially delivered to the system and then transferred from the CONVERSANT System to a live agent. The transferred call can be monitored with a VDN or ACD type of 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 of monitoring script that allows the call to be transferred to a nonmonitored domain or individual station. If the Data field of **A_Tran** was used to save voice script data, this data is made available in the CONVERSANT System 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 G3 System. Refer to “CONVERSANT System-to-Agent Transfers” below for additional design considerations.
- Incoming call directly to agent — In this scenario, monitored VDN’s or ACD splits deliver incoming trunk calls directly to live agents. Here, call events are passed to a VDN or ACD type of monitoring script and contain only ASAI-related information such as ANI, DNIS, and/or call-prompting information. Data screens are not based on data collected in a voice script since a CONVERSANT System voice script is not used to collect data from the caller. Since the CONVERSANT System does not service calls in this scenario, no data is present in the CONVERSANT System Data field of call events.
- Agent-to-agent transfers — In this scenario, calls are transferred between live agents. For example, *screening* agents can be used to collect information from the caller and handle simple transactions. The call can 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 other, specialized agents who can receive the call. The agent-to-agent transfer can be placed to a monitored domain or to an individual station and monitored with a VDN or ACD type of monitoring script. Note that the call may first have been delivered to the CONVERSANT System and then transferred to an agent prior to the live agent-to-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 CONVERSANT System Data can be present in call events. This information can be used in conjunction with data entered by a live agent to provide the basis for data screens. Refer to "Agent-to-Agent Transfers" below for additional design considerations.

Call-Flow Design

CONVERSANT System-to-Agent Transfers

CONVERSANT System-to-agent transfers are accomplished by using the **A_Tran** action within a voice script servicing a caller. The use of **A_Tran** invokes ASAI Third Party Call Control operations to transfer a call away from the T/R, LSE1, 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_Tran** action.

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

- The call can be transferred to a VDN or ACD split domain monitored by the CONVERSANT System 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 can be monitored using a CTL type monitoring script as described in Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. In this case, the call can be transferred to nonmonitored domains and individual stations. Here, only call events for calls transferred from the CONVERSANT System 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.

You can use a combination of the above two monitoring mechanisms on the same CONVERSANT System. Refer to Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for the rules for which monitoring script receives call events when these two mechanisms are combined

In addition to monitoring the transferred call, the application developer can save data collected in the voice script for subsequent use in the data screen delivery application. To do this, use the CONVERSANT System Data field of **A_Tran**. 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 that monitors the transferred call. The CONVERSANT System Data field provides storage for twenty characters. Note that multiple data items can be stored in this field. A social security number and PIN number, for example, can be collected in the voice script, concatenated, and then saved in the CONVERSANT System Data

field. The monitoring script that receives this data in call events can then unbundle the information for use in data screen delivery when the transferred call is delivered to an agent.

Typical Call Flow for CONVERSANT System-to-Agent Transfers

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

1. A call arrives at a T/R, LSE1, or LST1 channel on the CONVERSANT System. 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 CONVERSANT System Data field of **A_Tran**. The voice script terminates after the transfer is complete and the T/R, LSE1, 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 CONVERSANT System receives a CONNECT event for the call. The CONVERSANT System 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 for the transferred call is provided in a single CONNECT event.

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 CONVERSANT System Data information in addition to ASAI information related to the *original* call to the CONVERSANT System. The ANI and DNIS for the original call prior to the transfer, for example, are reported in this CONNECT event. Also, the Other Call ID field contains the call ID of the call originally delivered to the CONVERSANT System's T/R, LSE1, or LST1 channel. Call events for calls to T/R, LSE1, or LST1 channels on the CONVERSANT System 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 "Call-Flow Examples" below for a detailed call flow example.

CONVERSANT System-to-Agent Transfers Considerations

Additional considerations for CONVERSANT System-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 CONVERSANT System 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, LSE1, 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 can be transferred again 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 are discussed below in "Agent-to-Agent Transfers."
- The discussions on blind and consult transfers (refer to "Agent-to-Agent Transfers" below) do not apply to CONVERSANT System-to-agent transfers completed using the **A_Tran** action. Also, the delay needed for agent-to-agent transfers discussed later does not apply to CONVERSANT System-to-agent transfers completed using the **A_Tran** action.
- Transfers away from the CONVERSANT System can still be accomplished by using standard flash transfer mechanisms. This type of transfer, however, precludes the ability to use the CONVERSANT System 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 (refer to "Agent-to-Agent Transfers" below). Hence, the discussions on blind versus consult transfer and the need to introduce delay for blind transfers from the CONVERSANT System 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 G3 System.

Call Monitoring in Transfer Scenarios

The CONVERSANT System monitors VDN or ACD split domains by assigning monitoring scripts as described in Chapter 4, "Feature Packages," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. A call becomes monitored once it enters one of these monitored domains. The CONVERSANT System *must* also monitor all domains to which this call can be directed. Once monitored, therefore, a call remains monitored for the duration of the call even though it can 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 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.

Call ID Management in Transfers Scenarios

The DEFINITY G3 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 IDs 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 telephone.
3. A second call, temporarily independent of the first call, is placed from the transferring telephone. This call is assigned a call ID that is different from that of the original call. If this second call is placed to a monitored domain, the call immediately becomes monitored by the CONVERSANT System and call events can 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 telephone from the resultant call.

The CONVERSANT System is informed about the completed transfer immediately after the merge that occurs in Step 4. It is only after this merge, therefore, that the CONVERSANT System 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 CONVERSANT System can include information in the CONNECT event for the merged call which relates to the original call. In particular, the CONVERSANT System retains the call ID of the original call and reports it in the Other Call ID of

call events for the transferred call. This mechanism allows the host application to use call ID 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 CONVERSANT System 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 to associate the original call with the second call. The alternate mechanism is the CPN 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 CONVERSANT System 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 who can either receive calls via a monitored VDN or ACD split or via 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 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 "Call Flow Examples" below for detailed examples that include complete descriptions of call flows and call-event contents.

The following conditions should be noted for blind transfers:

- The domain receiving the original call and any domains receiving the transferred call must be monitored.
- 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 communications system can recognize the completion of the transfer 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 that has a “wait” step in it. The vector directs 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 for the CONNECT event to be passed to the script. The CONNECT event therefore only becomes available to the host application when the agent pushes the Transfer button the second time.
- In call-center operations that use blind transfer, the host application can 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.
- If for some reason calls are transferred to nonmonitored domains, unexpected operation can result. When the call placed by Agent 1 is not initially monitored, the CONVERSANT System assumes that a transfer to a station is taking place. Hence, two CONNECT events for the transferred call would be generated. One CONNECT event is generated when the transfer is completed by Agent 1 and another is 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 identifies the ACD split or VDN extension dialed by Agent 1, rather than identifying the extension of Agent 2. Note that the Connected Party Number field of the second CONNECT event for the transferred call identifies 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 can be transferred again after it is 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 case of a single transfer, 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 CONVERSANT System and is not 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 who 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 "Call Flow Examples" below for detailed examples that include complete descriptions of call flows and call-event contents.

The following conditions should be noted for consult transfers:

- 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 CONVERSANT System cannot assume that the two calls will eventually be merged since in some cases, they will not be. Hence, the two calls cannot be correlated by using call ID 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 can 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 Agent 1 presses the Transfer button the second time. 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 CONVERSANT System, 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, contains 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 nonmonitored domains, an unexpected operation can result. When the call to Agent 2 from Agent 1 is not initially monitored, the CONVERSANT System 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 identifies the ACD split or VDN extension dialed by Agent 1, rather than the extension of Agent 2.
- The END event 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 CONVERSANT System). 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 G3 entirely.
- The call can be transferred again after it is 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 case of a single transfer, 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 CONVERSANT ASAI System is integrated with a host computer. An application must be provided on the host that works with the CONVERSANT ASAI System. This host software application is not part of the CONVERSANT ASAI product. The host application can use the information it receives from the CONVERSANT ASAI System to do certain functions such as display call information on agent screens or route calls. The host application can also be called upon to provide the basis for an automated voice response application.

In some cases, particularly for voice response applications, the CONVERSANT ASAI System integrates well with an embedded application and hence no changes are required. For routing and data screen delivery applications, however, you will probably have to modify an existing application or provide a new one 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 CONVERSANT ASAI System. This is typically done by the customer's data-processing or information-systems department. Alternatively, you can purchase a third-party software vendor application that is designed to work with the CONVERSANT 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 CONVERSANT ASAI System, ISDN services, and any additional communications system ACD features.

The voice response, routing, and data screen delivery applications enabled by a CONVERSANT 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 telephone 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 PBX will probably have more than one ANI assignment. This is 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 CONVERSANT System T/R, LSE1, 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 G3 to the CONVERSANT System.
- 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 application involves transfers to live agents, refer to "CONVERSANT System-to-Agent Transfers" above 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 CONVERSANT 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 probably 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 CONVERSANT System 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 host must use the call events provided by a monitoring script on the CONVERSANT System to drive the screens on agent's terminals. The interface to the system 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 telephones 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 CONVERSANT System only at this point. The agent's telephone will alert the agent to the new call, and the agent can quickly finish work on the previous call. You may want to provide a quick way for the agent to move to this place in the interactions with the application.

- In data screen delivery applications, telephone extensions are used to identify agents receiving calls. The host application must therefore be able to associate extensions with particular data terminals. There are three possible ways to do this:
  - The agent can be queried for the telephone extension when the application is started. This is the most flexible arrangement and handles the situation where data terminals and terminal IDs are not permanently associated with the same telephone. Agents do make mistakes in providing the telephone extension to the system. You should plan for these occasional mistakes and make sure agents understand how to use the system properly. 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, the same extension, the extension information could be added to an agent profile.
  - If the relationship between data terminals, terminal IDs, and telephones is relatively stable, administration of the host application can maintain a fixed mapping between telephones and terminals.
- The agent screen application should be able to operate even if the CONVERSANT System 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 G3 continues to deliver calls to agents even if the ASAI link to the CONVERSANT System is down.
- If your call center application involves data screen delivery for CONVERSANT System-to-agent transfers, refer to "CONVERSANT System-to-Agent Transfers" on page 4-27 for additional design considerations.
- If your call center application involves data screen delivery for agent-to-agent transfers, refer to "Agent-to-Agent Transfers" on page 4-29 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 telephone and then at the telephone 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.

ASAI Application Examples

This section provides examples of scripts developed using the ASAI feature on the CONVERSANT System. Included in this section 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
- A monitoring script developed with the **A_Event** action

Sample ASAI Voice Script

The following is an example of an ASAI voice script 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

The following is an example of an ASAI routing script 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, an "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
        End External Action
        #
        # 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

The following is an example of an ASAI monitoring script 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
        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.
       #
4.     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
5.     Get Host Screen
       For Screen Name: verify
       End Get Host Screen
6.     Send Host Screen
       Send Screen Name: verify Use Aid Key: PF3
7.     Get Host Screen
       For Screen Name: connect_data
       End Get Host Screen
       #
       # Repeat the process - go back and get the next event.
       #
8.     Goto Event_start
       HOST_DOWN:
```

```
9.     Goto start
```

Call Flow Examples

This section provides examples of data screen delivery call flows and the contents of the call events that result from these call flows. The following call flows are described:

- Call to agent via ACD split
- Call to agent via VDNs with call-prompting
- Call to VDN and abandoned in queue
- Call to VDN and abandoned after agent selection
- Agent-to-agent transfer via VDN and blind transfer
- Agent-to-agent transfer to a station via blind transfer
- Agent-to-agent transfer via VDN and consult transfer
- Agent-to-agent transfer to a station via consult transfer
- CONVERSANT System-to-agent transfer via ACD split

In all call-flow scenarios, it is assumed that CONNECT events are triggered on ASAI *alerting* event reports. Hence, as shown in the scenarios, a CONNECT event is passed to a monitoring script when an agent is selected for a monitored call. An agent is considered to have been selected for a call when the agent's telephone begins ringing or the agent hears a zip tone. CONNECT events may also be triggered on ASAI *connected* event reports. In this case, CONNECT events are passed to monitoring scripts when agents actually answer monitored calls.

In all call flow scenarios it is assumed that the incoming call is delivered via an ISDN facility. This implies that the ANI is available in the ISDN SETUP message for the incoming call. If ANI is available, it is reported in call events as depicted in the call flow scenarios. If ANI is not available, then the incoming trunk group ID would be reported instead.

Also, since it is assumed that incoming calls are delivered via an ISDN facility, a 10-digit CPN is reported in call events. This number corresponds to the Called Party Number provided in the ISDN SETUP message for the incoming call. Note that, as depicted in the call-flow scenarios, this number identifies a billing number and not the 800 number dialed by the caller. The use of switch administration to modify DNIS digits does not affect the reporting of the called party number for incoming ISDN calls.

Incoming calls can also be delivered via non-ISDN facilities. In this case, ANI is not available, so the trunk group ID would always be reported instead. Also, for non-ISDN calls, the called party number identifies the ACD split or VDN extension to which the call is initially directed. Hence, for non-ISDN calls, the use

of switch administration to modify DNIS digits can affect the reporting of the CPN. If modified by switch administration, the DNIS digits, as provided by the network, are not reported in the called party number. Rather, the ACD split or VDN extension that results from the modification is provided in the CPN.

Scenarios 6 through 9 discuss agent-to-agent *transfer* calls. Note that the call events generated for agent-to-agent *conference* calls are the same as described in the transfer scenarios. The three functional differences for conference calls are:

- The screening agent uses the Conference button instead of the Transfer button.
- The screening agent stays on the call instead of being dropped off.
- The END event for the call is not generated until all parties disconnect from the call.

Call to an Agent via an ACD Split

A call arrives at the DEFINITY G3 System and is delivered directly to a monitored ACD split (no vector processing takes place for the call). An agent is assigned to the call, interacts with the caller, and then terminates the call.

Example:

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service.

   Calls to that 800 number are presented to a monitored ACD split with the extension 7777.

2. The call is queued to the monitored ACD split 7777.
3. The call is assigned to an agent in that split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for ACD split 7777 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557777
Switch Data	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. When the selected agent completes and disconnects the call, an END event is passed to the monitoring script for ACD split 7777 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557777
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Call to an Agent via VDNs with Call Prompting

A call arrives at the DEFINITY G3 System and is handled with call vectoring. The initial VDN/vector that processes the call makes use of the call-prompting feature on the DEFINITY G3 System to collect information from the caller. In particular, the caller is asked to request a service, for example, “press 1 for gizmo service or press 2 for widget service.” The call is then routed unconditionally to a second VDN that is monitored. The vector associated with the second VDN queues the call to an ACD split. Agents in this split can handle service calls for both products. The call-prompting information collected on the DEFINITY G3 System can be used to determine which application to start up when the call is delivered to an agent in the common agent group. This allows a single 800 number to be advertised for both products.

Example:

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are initially handled with a vector associated with VDN 7771. VDN 7771 is not monitored. This vector prompts the user to enter a 1 or 2 and then routes the call to VDN 7772 with a “route to” step. In this example it is assumed that the caller inputs a 1.
2. The call is routed to the monitored VDN 7772. The vector associated with VDN 7772 queues the call to an ACD split with a “queue to” step.
3. The call is assigned to an agent in the split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for VDN 7772 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	1
Trunk Group Id	

Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. When the selected agent completes and disconnects the call, an END event is passed to the monitoring script for VDN 7772 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	1
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Call to a VDN and Abandoned in Queue

A call arrives at the DEFINITY G3 System and is handled with a VDN or vector. The vector queues the call to an ACD split. The caller abandons the call while it is in the queue and before it is assigned to an agent.

Example:

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed by a vector associated with VDN 7771. VDN 7771 is monitored.
2. The vector associated with VDN 7771 queues the call to a vector-controlled split with a “queue to” step.
3. The caller abandons before an agent is assigned to the call.
4. An ABANDON event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	

VIS Data	
Routing ID	
Return Field	A

Call to a VDN and Abandoned After Agent Selection

A call arrives at the DEFINITY G3 System and is handled with a VDN or vector. The vector queues the call to an ACD split. The caller abandons the call after is was assigned to an agent but before the agent could answer it.

Example:

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed by a vector associated with VDN 7771. VDN 7771 is monitored.
2. The vector associated with VDN 7771 queues the call to an ACD split with a “queue to” step.
3. An agent at extension 1234 is selected for the call.
4. The caller abandons the call before the agent at extension 1234 can answer.
5. An ABANDON event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	A

Note that this is different from the previous scenario where the caller abandons the call while in the queue. Since an agent was selected for the call before it was abandoned, a CONNECT event is passed to the monitoring script. In the previous case where the caller abandons the call while it is in the queue, no agent was selected for the call; therefore, no CONNECT event is passed to the monitoring script before the ABANDON event. In this scenario, where the caller abandons the call after agent selection, the ABANDON event contains the extension of the agent selected for the call. This information can be used to cancel the CONNECT event for the call to the agent since the call terminates before the agent can interact with the caller. Alternatively, the host application could simply let the next CONNECT event for the same agent “overwrite” the

previous CONNECT for the call that was abandoned. The next CONNECT event comes when the next call is delivered to the agent.

Note also that this scenario only applies when CONNECT events are triggered on ASAI alerting event reports. If CONNECT events are triggered on ASAI connect event reports, CONNECT events are passed to monitoring scripts only when agents actually answer calls. Consequently, for cases where CONNECT events are triggered on ASAI connect event reports, only an abandon while in the queue situation is possible. An abandon after agent selection situation will never occur or be reported.

Agent-to-Agent Transfer via a VDN and Blind Transfer

A call is delivered to an agent within a screening split. The screening agent transfers the call using a blind transfer to a group of specialized agents. A delay is built into the transfer by having the screening agent place the transfer call to a VDN. The vector associated with this VDN queues the call to the specialized agent group after delaying the call. This delay allows the transfer to be completed prior to when the transfer call is delivered to a specialized agent.

Example

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed with a vector associated with VDN 7771. VDN 7771 is monitored.
2. The vector associated with VDN 7771 queues the call to the split of screening agents.
3. The call is assigned to an agent in the screening split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. The screening agent talks with the caller and determines that a transfer is necessary.

6. The screening agent at extension 1234 presses the Transfer button and dials 7770, the extension of a monitored VDN.
7. The vector associated with VDN 7770 delays the call placed by the agent at extension 1234 for 2 seconds with a “wait” step.
8. The screening agent at extension 1234 then presses the Transfer button again. This merges the two calls at the screening agent’s telephone and drops the screening agent from the resultant call. Note that no END event is reported at this time.
9. The vector associated with VDN 7770 queues the resultant call to the group of specialized agents with a “queue to” step.
10. A specialized agent at extension 4681 is selected for the transferred call.
11. A CONNECT event is passed to the monitoring script for VDN 7770 with the following information:

Connected Party Number	4681
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

12. The specialized agent at 4681 completes the call and disconnects.
13. An END event is passed to the monitoring script for VDN 7770 with the following information:

Connected Party Number	4681
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Note that for blind transfers to monitored domains as described in this scenario, the second CONNECT event identifies the original call in the Other Call Id field. Note also that this CONNECT event contains ASAI information that pertains to the original call (for example, original ANI and DNIS in the Calling Party Number and Called Party Number fields, respectively). Any LAI display information, VIS data, or switch data associated with the original call is carried forward as well.

Agent-to-Agent Transfer to a Station via Blind Transfer

A call is delivered to an agent within a screening split. The screening agent transfers the call using blind transfer to a specialized agent at an individual station.

Example

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed with a vector associated with VDN 7771. VDN 7771 is monitored.
2. The vector associated with VDN 7771 queues the call to the split of screening agents.
3. The call is assigned to an agent in the screening split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. The screening agent talks with the caller and determines that a transfer is necessary.
6. The screening agent at extension 1234 presses the Transfer button and dials 2022. Extension 2022 identifies an individual station associated with a single, specialized agent.
7. The call initiated by the agent at extension 1234 begins ringing at station 2022. Note that no CONNECT event is reported for this call at this time since the CONVERSANT System is not yet monitoring this call.
8. The screening agent at extension 1234 then presses the Transfer button again. This merges the two calls at the screening agent's telephone and drops the screening agent from the resultant call. Note that no END event is reported at this time.

9. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	2022
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

   Note that this CONNECT event for blind transfers to stations is not passed to a monitoring script until the screening agent completes the transfer by pressing the Transfer button a second time.

10. The specialized agent at 2022 answers the transferred call and begins interacting with the original caller.
11. The specialized agent at 2022 completes the call and disconnects.
12. An END event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	2022
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Note that for blind transfers to stations as described in this scenario, the second CONNECT event identifies the original call in the Other Call Id field. Note also that this CONNECT event contains ASAI information that pertains to the original call (for example, original ANI and DNIS in the Calling Party Number and Called Party Number fields, respectively). Any LAI display information, VIS data, or switch data associated with the original call is carried forward as well.

Agent-to-Agent Transfer via a VDN and Consult Transfer

A call is delivered to an agent within a screening split. The screening agent transfers the call using consult transfer to a group of specialized agents.

Example

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed with a vector associated with VDN 7771. VDN 7771 is monitored.
2. The vector associated with VDN 7771 queues the call to the split of screening agents.
3. The call is assigned to an agent in the screening split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. The screening agent talks with the caller and determines that a transfer is necessary.
6. The screening agent at extension 1234 presses the Transfer button and dials 7772, the extension of a monitored VDN.
7. The vector associated with VDN 7772 queues the call to the group of specialized agents.
8. A specialized agent at extension 4440 is selected for the call placed by the agent at extension 1234.
9. A CONNECT event is passed to the monitoring script for VDN 7772 with the following information:

Connected Party Number	4440
Calling Party Number	1234
Called Party Number	7772
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

10. The screening agent and the specialized agent talk privately while the original caller is on hold.
11. The screening agent at extension 1234 then presses the Transfer button again. This merges the two calls at the screening agent's phone and drops the screening agent from the resultant call. The original caller is now connected to the specialized agent at extension 4440. Note that no END event is reported at this time.
12. The specialized agent at extension 4440 interacts with the original caller.
13. The specialized agent at 4440 completes the call and disconnects.
14. An END event is passed to the monitoring script for VDN 7772 with the following information:

Connected Party Number	4440
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Note that for consult transfers to monitored domains as described in this scenario, the second CONNECT event does not identify the original call in the Other Call Id field. Note also that this CONNECT event does not contain ASAI information that pertains to the original call. Only call events passed to a monitoring script after the transfer is completed contain this information (for example, the END event or a CONNECT event for a subsequent blind transfer). Any LAI display information, CONVERSANT System data, or switch data associated with the original call is carried forward as well and reported in call events reported after the transfer is complete.

Agent-to-Agent Transfer to a Station via a Consult Transfer

A call is delivered to an agent within a screening split. The screening agent transfers the call using consult transfer to a specialized agent at an individual station.

Example

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed with a vector associated with VDN 7771. VDN 7771 is monitored.

2. The vector associated with VDN 7771 queues the call to the split of screening agents.
3. The call is assigned to an agent in the screening split with the extension 1234.
4. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	101
Other Call Id	
LAI Display Info	
VIS Data	
Routing ID	
Return Field	C

5. The screening agent talks with the caller and determines that a transfer is necessary.
6. The screening agent at extension 1234 presses the Transfer button and dials 2022. Extension 2022 identifies an individual station associated with a single, specialized agent.
7. The second call initiated by the agent at extension 1234 begins ringing at station 2022. Note that no CONNECT event is reported for this call at this time since the CONVERSANT System is not yet monitoring this call.
8. The specialized agent at extension 2022 answers the call from the screening agent at extension 1234.
9. The screening agent and the specialized agent talk in a private conversation while the original caller is on hold.
10. The screening agent at extension 1234 then presses the Transfer button again. This merges the two calls at the screening agent's phone and drops the screening agent from the resultant call. The original caller is now connected to the specialized agent at extension 2022. Note that no END event is reported at this time.
11. A CONNECT event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	2022
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	

VIS Data	
Routing ID	
Return Field	C

Note that this CONNECT event for consult transfers to stations is not passed to a monitoring script until the screening agent completes the transfer by pressing the Transfer button a second time.

12. The specialized agent at 2022 interacts with the original caller.
13. The specialized agent at 2022 completes the call and disconnects.
14. An END event is passed to the monitoring script for VDN 7771 with the following information:

Connected Party Number	2022
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	
Routing ID	
Return Field	E

Note that for consult transfers to stations as described in this scenario, the second CONNECT event identifies the original call in the Other Call Id field. Note also that this CONNECT event contains ASAI information that pertains to the original call (for example, original ANI and DNIS in the Calling Party Number and Called Party Number fields, respectively). Any LAI display information, CONVERSANT System data, or switch data associated with the original call is carried forward as well.

CONVERSANT System-to-Agent Transfer Via an ACD Split

A call is delivered to a CONVERSANT System T/R, LSE1, or LST1 channel and serviced by an ASAI voice-response application. An account number is collected in the voice script in preparation for a data screen delivery application based on this account number. The call is then transferred to a live agent group.

Example:

1. A caller calling from the telephone number 303-555-1726 calls a toll-free 800 number associated with customer service. Calls to that 800 number are processed with a vector associated with VDN 7771. VDN 7771 is not monitored.

2. The vector associated with VDN 7771 routes the call to the CONVERSANT System T/R, LSE1, or LST1 split with a “route to” step. Note that this T/R, LSE1, or LST1 split is monitored but not by a monitoring script used to retrieve call events. This split is monitored internally by the CONVERSANT System to support ASAI voice response applications making use of the **A_Callinfo** and **A_Tran** actions.
3. The call is answered by a T/R, LSE1, or LST1 channel and serviced by a voice response script. No CONNECT event is passed to a monitoring script for the call at this point. Assume, however, that this call is assigned call ID 101. This call ID would be available within the voice script by using the **A_Callinfo** action.
4. The voice script collects an account number from the caller. In this example, it is assumed that the account number is 987654321.
5. The A_Tran action is used within the voice script to transfer the call to the monitored ACD split 7777. The Destination Number field of **A_Tran** is set to 7777 and the CONVERSANT System Data field of **A_Tran** is set to 987654321.
6. When the transfer is executed, the voice script terminates allowing the T/R, LSE1, or LST1 channel to service additional calls.
7. The call queues to the monitored ACD split 7777.
8. An agent at extension 1234 within ACD split 7777 is selected for the call.
9. A CONNECT event is passed to the monitoring script for ACD split 7777 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105
Other Call Id	101
LAI Display Info	
VIS Data	987654321
Routing ID	
Return Field	C

10. The agent at extension 1234 answers the call and interacts with the caller.
11. The agent at extension 1234 completes the call and disconnects.
12. An END event is passed to the monitoring script for ACD split 7777 with the following information:

Connected Party Number	1234
Calling Party Number	3035551726
Called Party Number	9085557771
Switch Data	
Trunk Group Id	
Call Id	105

Other Call Id	101
LAI Display Info	
VIS Data	987654321
Routing ID	
Return Field	E

Note that for CONVERSANT System-to-agent transfers as described in this scenario, only one CONNECT event is reported to a monitoring script. This CONNECT event is reported when a live agent answers the transferred call. Not also that this CONNECT event contains data in the CONVERSANT System Data field if such data was saved in the voice script via the use of the **A_Tran** action. The CONNECT event also identifies the original call in the Other Call Id field and contains ASAI information that pertains to the original call (for example, original ANI and DNIS in the Calling Party Number and Called Party Number fields, respectively). Any LAI display information or switch data associated with the original call is carried forward as well.

Note that the call may also be transferred from the CONVERSANT System to a nonmonitored domain or individual station. In this case the call events are the same as those described in this scenario. The call events, however, are passed to a CTL-type monitoring script instead of a VDN- or ACD-split-type monitoring script. Also, A_Tran must be used to ensure that the CTL-type monitoring script receives the call events for the transferred call. Refer to Chapter 8, “Using Optional Features,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760, for additional information.

Converse Vector Step Routing

5

What's in This Chapter

This chapter describes the use of the Converse Vector Step (CVS) and the requirements that must be met to implement this interface. It also provides a list of application development issues that you must address when using the CVS.

What is the Converse Vector Step?

The Converse Vector Step (CVS) allows the PBX to maintain control of a call while capabilities of the INTUITY CONVERSANT Voice Information System are being used. To facilitate this control, the Script Builder conv_data external action supports the DEFINITY G3V2 Voice Response Integration feature (load 04.2.0.096 or greater) on Tip/Ring (T/R) and Line Side T1 and E1 lines.

Without the use of the CVS, once the call terminates on the CONVERSANT System channel, it is no longer under the control of the switch. The system must process the transaction further and then route the response back to the switch using the Transfer Call action. With the CVS, control over call-routing is retained by the switch.

At the beginning of the script, the CVS allows touch tones to be passed to the CONVERSANT System containing information such as the Automatic Number Identification (ANI). At the end of the script, the system can pass back information relevant to further call vectoring via touch tones, such as a customer account number.

For additional information about the DEFINITY Voice Response Integration feature, refer to *DEFINITY Communications System Generic 3 Call Vectoring and Expert Agent Selection (EAS) Guide*, 555-230-620.

CVS Provisioning

The following information details the necessary provisioning for the CVS on the CONVERSANT System and the PBX.

Provisioning Within the CONVERSANT System

The Converse Data Return (conv_data) action can only be implemented on the T/R, LST1, and LSE1 channels. Therefore, the application to be used must be assigned to the appropriate T/R, LST1, and LSE1 channels. Refer to the hardware installation guide for your platform for information about installing the necessary circuit cards. Refer to Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for procedures on assigning service to channels.

The conv_data external action is a part of the base software (V4 and later). The INTUITY CONVERSANT System's base software must be installed prior to implementation of the CVS. Look under "Installing the Base System Software," of the appropriate Intuity ™ CONVERSANT® System Version 6.0 maintenance book.

Provisioning within the PBX

The Converse Data Return (conv_data) action can only be used when the INTUITY CONVERSANT System is used with the DEFINITY switch containing DS1 cards (version 767D or later in switch V5 or later). You must verify the G3V2 switch load prior to implementing the CVS. Failures occur in feature operation unless the G3V2 switch is running load 04.2.0.096 or greater.

CVS Administration

The following information details the necessary administration of the CVS on the CONVERSANT System and the PBX.

Administration Within the CONVERSANT System

The conv_data return action executes a flash, and then transmits the digits contained in the Feature Access Code (FAC) and Data Return fields for conv_data. The duration of this flash must be set on the CONVERSANT System at 600 msec in the Switch Hook Flash Duration field. If you are using T/R lines,

set this value in the Analog Interfaces screen. If you are using LST1 lines, set this value in the Digital Protocol: Line Side T1 - DEFINITY screen. Refer to Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

Set the Dial Tone Delay in the Digital Protocol: Line Side T1 - DEFINITY screen. Try values of between 200 and 1000 msec. Even a 1000 msec value may not be sufficient for dial-tone delay. Dial tone may not occur if your switch does not have enough dial-tone detection registers. If you think that lack of dial tone may be a problem, extend this value. Unfortunately, this may cause additional delays in the data return phase and the customer may hear dead air on the line.

Administration within the PBX

If the Converse Data Return action step is implemented on LST1 and LSE1 channels, you must set the Converse First Data Delay parameter on the Systems Parameters Features screen on the DEFINITY to 1 instead of 0 (zero). (Zero is the default setting).

The Feature Access Code field in the conv_data action must match the corresponding FAC code setup on the switch. Refer to the DEFINITY G3V2 Call Vectoring documentation for more information.

CVS Application Development Issues

To use the CVS, you must

- Set up parameters to facilitate data-passing from the switch within the framework of the application being developed
- Define the data-return parameters to enable the CONVERSANT System to send the collected data back to the switch

Script Builder

The Prompt and Collect action step and the conv_data external action are available to use the CVS within an application. Refer to Chapter 7, “Defining the Transaction,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760.

To isolate and resolve problems during the CVS execution, application developers should use the call data event capabilities of Script Builder to log information about return code status. Refer to Chapter 6, “Defining Parameters,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760. This ensures that the Call Data Detail report reflects the outcome of the call using the DEFINITY feature. For example, the return code for conv_data can be stored in a variable and that variable can be one of the logged events in the Call Data Events screen.

Script Language

The **chantype** script instruction allows scripts to determine the type of channel on which they are running. Find more information under “Script Instructions,” and “Summary of Script Instructions,” in *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods* 585-310-761.

INTUITY Response Application Programming Interface

It is possible to write an INTUITY Response Application Programming Interface (IRAPI) application that is installed as a start-up service to collect calling party and/or called party information. This application sets the IRD_ANI and/or IRD_DNIS information elements before “exec’ing” the desired application via the number services tables. The *irDial()* and *irGetInput()* functions can be used to exchange data with the switch. For additional information about these capabilities, refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods*, 585-310-761.

CVS Versus ASAI

For a discussion of the relative benefits of CVS and ASAI, refer to "ASAI Versus Converse Vector Step" on page 4-16.

CVS Examples

The following are typical ways in which CONVERSANT System applications can use the CVS.

- Port sharing

  By specifying the vector directory number (VDN) in the <data1> field for the CVS on the DEFINITY switch, information equivalent to DNIS is available to the CONVERSANT System via the first Prompt & Collect action. Based on the VDN, the CONVERSANT System can execute an appropriate script using the Script Builder Execute action. This capability is similar to DNIS on T1 E&M channels. Prior to the CVS, T/R channels only had this capability through ASAI.

- Automatic number identification (ANI) – Called Party Number/Billed Number (CPN/BN)

  By specifying ANI in the <data2> field and VDN in the <data1> field for the CVS on the DEFINITY switch, the CPN/BN is available to the executed script via the second Prompt & Collection action. This information may be useful in a dealer or locator application.

- CONVERSANT System announcement selection

Hard-coded administered digit strings in the <data_1> and/or <data_2> fields can be used to instruct the CONVERSANT System to play selected announcements.

- Indication of anticipated delay

  If your CONVERSANT System is used with a Definity G3V4 or later, a callers expected wait time is passed using the keword *EWT*. If your switch is G3V2 or G3V3, a caller's position in queue is passed using the keyword *qpos*. The CONVERSANT System can play an announcement Informing the caller of an anticipated wait based on either data item.

- ANI/routing

  Based on the CPN/BN, the CONVERSANT System can perform a database operation to determine further routing of the call. For example, in a credit card application, the CPN/BN can map to a premier account holder or a regular account holder. This information can be passed back using the data return string so that the DEFINITY can give priority treatment as required. The account number can be directed to appear on the agent's display.

- Enhanced call management System (CMS) call records

  Digit strings passed back to the DEFINITY can be stored in CMS call records to provide further detail as to call dispositions (for example, the number of premier versus regular account calls processed by the CONVERSANT System).

Call Classification Analysis

6

What's in This Chapter

This chapter describes the use of CCA and the benefits it provides in analog and digital communications. It also details requirements for implementing this feature and suggested values for telephony parameters when using this feature.

What is CCA?

CCA allows application developers to classify the disposition of originated and transferred calls. There are two types of CCA:

- Intelligent — This type of call classification supports call transfers and call bridges. It uses the signalling and tone detection capabilities of the network interface card that is being used. Intelligent CCA is intended only for use on outbound calls that terminate on the switch or PBX to which the INTUITY CONVERSANT System is connected.
- Full — This type of call classification provides enhanced capabilities to intelligent call classification such as better answer detection, busy and audible ring tone detection, modem tone detection, etc. It is offered as an optional feature package. Full CCA should be used when outbound calls will terminate beyond the local switch or PBX.

  Full Call Classification Analysis (CCA) is used only in the United States of America and Canada.

CCA Provisioning

Full CCA requires at least one Signal Processor circuit card or one Speech and Signal Processor card to be installed and operational prior to loading the Full CCA software. A single Signal Processor (SP) card can handle 6 simultaneous channels of CCA. A single Speech and Signal Processor (SSP) card supports 42 simultaneous channels of CCA. In either case the (S)SP card must be dedicated to call classification (see "CCA Administration") and connected to the TDM bus. Refer to the appropriate hardware installation book for your platform for information on installing the (S)SP circuit card.

Intelligent CCA on T1 or Primary Rate Interface (PRI) digital lines provides answer and disconnect supervision only. Unless an AYC21 card provides your digital interface, intelligent CCA is not available on Line Side T1 (LST1) lines because there is no answer supervision or dial-tone detection.

If you require detection of call progress tones (different from dial tone detection) with LST1, you must either install Full CCA or install an AYC21 circuit card. If you require detection of call progress tones with T1 (E&M) or PRI, you must install Full CCA.

NOTE:
Unless an AYC21 card provides your digital interface, LST1 cannot detect dial tone or stutter dial tone prior to dialing whether or not it is used with the Full CCA feature.

NOTE:
CCA performance may be slightly less if used with analog T/R lines instead of digital lines. Analog lines tend to be more noisy than digital lines and may lead to occasional false identification of tones.

NOTE:
Since full CCA is designed only for the tone plan used in the US and Canada, full CCA is not recommended for use with E1 (CAS) or LSE1 (those protocols typically used outside the US and Canada).

CCA Administration

You must assign CCA functionality to the (S)SP circuit card for the CCA feature to operate properly. To assign CCA functionality to an (S)SP circuit card, refer to Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for the procedure to change the state of the (S)SP circuit card.

CCA Application Development Issues

The following covers general, Script Builder, script language and INTUITY Response Application Programming Interface (IRAPI) development issues with CCA.

General Issues

An error is generated if a script attempts to use Full CCA and the maximum number of CCA instances (six if you use an SP, 42 if you use an SSP) are running. No further attempts to use Full CCA are made after the error is logged. Refer to system message TSM003 under "System Message Listings," in the appropriate Intuity ™ CONVERSANT® System Version 6.0 maintenance book.

Script Builder

Intelligent CCA or Full CCA can be activated when a call is dialed out during a flash transfer, a call bridge (internal transfer), or a make call (call origination), as defined in Script Builder. The Script Builder actions: Transfer Call, Call Bridge, and Make Call use intelligent and Full CCA. Refer to *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder*, 585-310-760, for additional information.

NOTE:
You must use the Make Call, Transfer Call, and Call Bridge actions to populate the database used in generating the Call Classification Report. The Call Classification Report provides information for each extension or number dialed, the total number of calls, and the number of transfer attempts for a specified date. Refer to Chapter 5, "Reports," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for information about the Call Classification Report.

Script Language

The following instructions invoke Full CAA through script language:

- **setcca**
- **tic**

The information below gives a brief discussion of these two instructions. For detailed information, look under, "Script Instructions," and "Summary of Script Instructions," of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods* 585-310-761

setcca

The **setcca** script instruction allows the application developer to set CCA parameters at the script level. These parameters specify the following:

- Whether to use intelligent or Full CCA
- The number of rings to wait for an answer
- Whether to use answer detection or speech-energy detection

tic

The **tic** instruction specifies additional call dispositions for Full CCA if Full CCA is turned on via the **setcca** instruction.

IRAPI

The *irSetParam(3irAPI)* function can be used to set the IRP_OUTCALL_CCALEVEL to IRD_FULL_CCA. This parameter enables Full CCA on a channel for subsequent *irCall(3irAPI)* and *irDial(3irAPI)* function calls.

CCA Example

The following example is an excerpt from a script showing how an application developer might use the **setcca** and **tic** instructions in a Full CCA application.

```
setcca(im.1,im.10,im.-1)
nextcall:
dbase( .... )     /* get number to dial from DIP */

tic('O', r.3)    /* call number in register 3 */

jmp(r.0 == im.'N', noAns)     /* no answer after 10 rings */
jmp(r.0 == im.'B', busy)
jmp(r.0 == im.'F', retry)
jmp(r.0 == im.'A', answer)
jmp(r.0 == im.'s', SIT)
jmp(r.0 == im.-4, noResource)

noAns:
tic('h')     /* put line on-hook to stop ringing */

busy:
dbase ( .... )    /* report result to controlling DIP */
goto (nextcall)

SIT:
jmp(r.1 == im.'R', retry)
jmp(r.1 == im.'r', retry)
jmp(r.1 == im.'K', retry)
```

```
jmp(r.1 == im.'k', retry)
dbase ( .... )     /* report result to controlling DIP */

answer:
talk("Hello, you may be the winner of a free trip to Hawaii")
dbase ( .... )     /* report result to controlling DIP */
goto (nextcall)
```

Data Network Communications

7

What's in This Chapter

The following data network communication interfaces are available for use in conjunction with the INTUITY CONVERSANT Voice Information System (VIS) V6.0 software.

- SNA 3270
- TCP/IP
- SQL*NET
- Asynchronous

This chapter provides information on each of these packages, including the configuration and administration procedures.

Host Interface Software

The host interface is a combination of hardware and software designed to allow the transmission of information over a network. This network usually includes remote host computers and/or databases. The Host Interface software package allows applications running on the VIS to send and receive screens from applications running on the host mainframe. This offer was created in conjunction with CLEO Communications.

Host Interface Architecture Overview

The host interface software emulates an IBM 3274-41C or a 3174-01R cluster controller with up to 128 logical units (LUs) (that is, 3278 Model 2 terminals) connected to it. The host interface card (either a PC/XL, FIFO/SIB, or token ring circuit card) is typically linked to a 3725 or 3745 Front End Processor (FEP) and uses synchronous data link control (SDLC) or token ring data streams.

Hardware Environment Architecture

Figure 7-1, Figure 7-2, and Figure 7-3 illustrate typical VIS-to-host connections using modems, a modem eliminator, or a token ring.

Standard links from the host interface card to the FEP can be made through synchronous modems for distances over 30 meters (100 feet) and leased lines or modem eliminators for distances under 30 meters (100 feet). The software supports speeds up to 56-Kbyte baud with the following restrictions:

- Certain line configurations are required to operate at speeds higher than 9600 baud. Refer to the information provided later in this section for assistance in modem configurations.
- High-speed connections for 56-Kbyte baud operation may use modems or modem eliminators with V.35 connectors. This requires an RS-232-to-V.35 interface converter since the PC/XL and FIFO/SIB circuit cards have only an RS-232 connector.
- If you have a PC/XL and want to support 56-Kbyte baud operations, you must have a Revision D or later version of the PC/XL card. To locate the revision letter of your card, look on the back side of the card near the RS232 connector. If your card is an earlier version than Revision D, you must upgrade to the FIFO/SIB circuit card.

NOTE:
The PC/XL circuit card is not an orderable replacement for CONVERSANT VIS V6.0. If a replacement SDLC card is needed, you must order the FIFO/SIB circuit card. Refer to the hardware installation book for your platform for additional information.

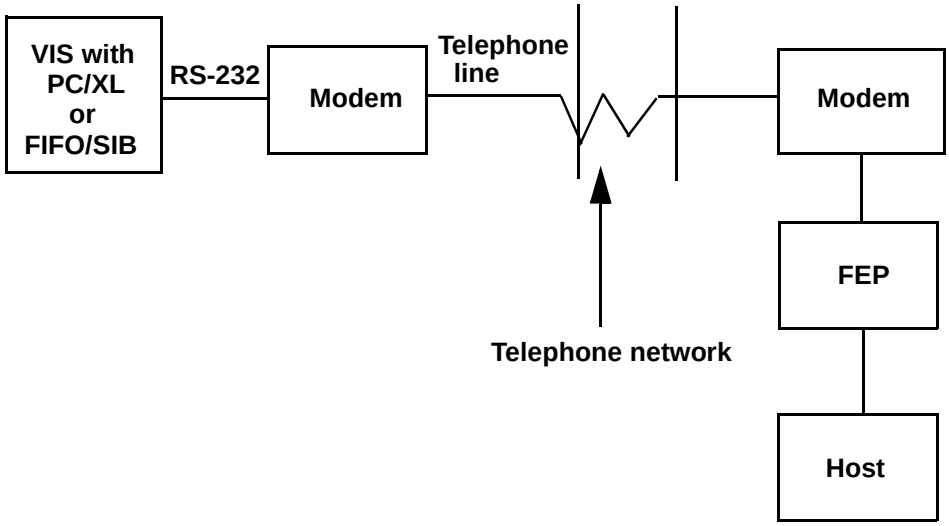


Figure 7-1. Sample Host Connection Using a Modem

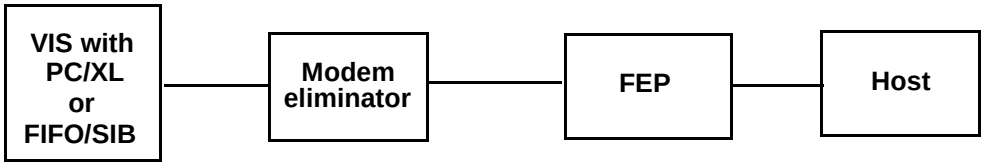


Figure 7-2. Sample Host Connection Using a Modem Eliminator

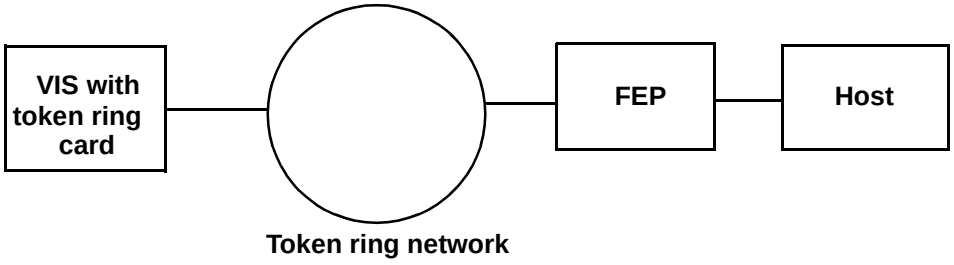


Figure 7-3. Sample Host Connection Using a Token Ring

The following list details the possible configurations for the host interface cards that are supported in a single platform for INTUITY CONVERSANT V6.0. Note that a total of three host connections is only possible when using two FIFO/SIB cards and a Token/Ring card.

- 1 or 2 FIFO/SIB card(s)
- 1 PC/XL card
- 1 token ring card
- 1 or 2 FIFO/SIB card(s) and 1 token ring card
- 1 PC/XL card and 1 token ring card

Software Process Architecture

In INTUITY CONVERSANT System Version 6.0 software, CLEO's LINKix product provides the Systems Network Architecture (SNA) software, including 3270, SNA and link levels. Two link-level protocols are supported:

- Token ring — A ring type of local area network (LAN) that allows any station in the network to communicate with any other station
- Synchronous Data Link Control (SDLC) — A line protocol that supports point-to-point communication

NOTE:
The token ring protocol has a higher throughput than the SDLC line protocol.

The following figure shows the current software process architecture for the host interface. Note that in Figure 7-4 the dashed line separates the process ownership between the VIS and LINKix.

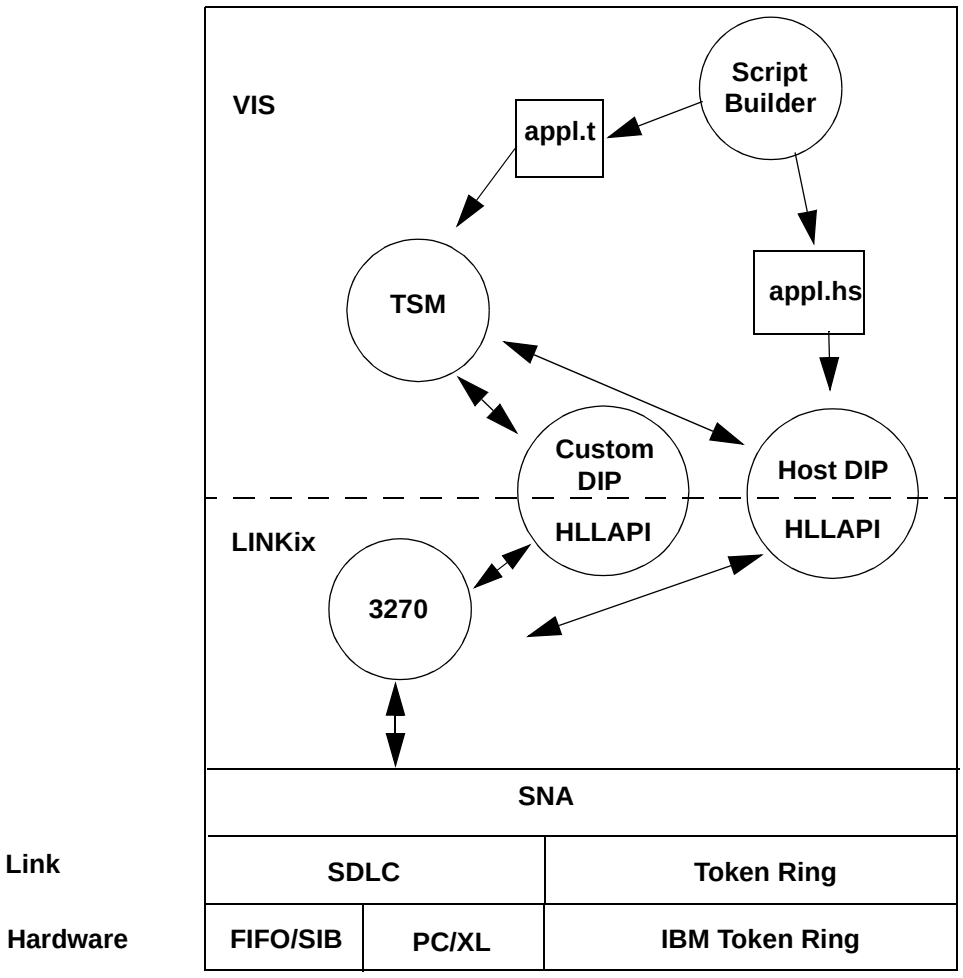


Figure 7-4. Host Interface Protocol

Host Interface Features

The following are the basic features of the Host Interface software available with INTUITY CONVERSANT VIS V6.0 and CLEO Communications software:

- Script Builder applications interface with host programs

  Script Builder can be used to create an application to interface with a complicated host computer application. The application developer logs on to the host computer and captures screen images, then identifies the screens and fields on those screens that are needed during the transaction. An external function can also be created to allow Script

Builder to interface with custom data interface processes (DIPs) that require data communications protocols other than 3270 SNA or the High Level Language Application Programming Interface (HLLAPI),

- 3270 Terminal Emulation

  This capability allows a device or program to imitate another device or program. The 3270 terminal emulation software temporarily transforms itself into a look-alike of an IBM 3270 terminal. In addition to providing full 3270 functionality, the 3270 Terminal Emulator allows the transfer of files to and from UNIX.

- IND$FILE File Transfer

  The file transfer capability allows you to transfer text or binary files between a mainframe using the IBM host program IND$FILE and your INTUITY CONVERSANT VIS.

  NOTE:
  In past CONVERSANT VIS releases, this file transfer capability was a part of the 3270 Host Communication Package File Transfer System (FTS). It is now a part of the basic Host Interface software offer.

  FTS can work with multiple IBM mainframe operating environments or processing subsystems. These host systems and their IBM IND$FILE program product numbers include:

  — Time Sharing Option (TSO), #5665-311

  — Conversational Monitor System (CMS), #5663-281

  — Customer Information Control System (CICS), #5789-DQH

  Once installed, file transfer can be initiated interactively through the Terminal Emulator (TE) program or directly from the UNIX command line either by entering the FTS commands or by running a shell script containing the commands.

  NOTE:
  The host configuration for file transfer must use the write structured fields (WSF), also known as DFT mode, in the logmode table entry. This entry must be verified in both upgrades and new systems as this is a change from previous product releases.

- Enhanced File Transfer

  Enhanced File Transfer uses the file transfer system to automatically transfer files between the VIS and a synchronous host processor on a designated LU.

- HLLAPI

  HLLAPI is an application programming interface that allows users to write custom applications that can communicate with the host via an API. Refer to the *HLLAPI Programmer's Guide* for additional information.

- NetView

  NetView provides the capability to send error messages to the host as Operator Generated Alerts (OGAs). LINKix also provides a more general NetView interface allowing messages to be sent to the host as other types of alerts. However, the VIS NetView package (see "Software Package Structure") does not use this more general interface.

Software Package Structure

The following packages comprise the Host Interface software for Version 6.0. Note that the Link, SNA, and Feature Levels are owned by CLEO Communications. Refer to the appropriate *Intuity* ™ CONVERSANT® System Version 6.0 maintenance book, for information about the order in which you must install these packages.

- Link Levels — The link level package(s) you need depends on the type of protocol and the type of interface card(s) being used in the configuration.
  - — linkix_coproc, Link Level (coproc). This package contains the CLEO LINKix PC/XL SDLC Link Level.
  - — linkix_sib, Link Level (sib). This package contains the CLEO LINKix FIFO/SIB SDLC Link Level.
  - — linkix_tkrn, Link Level (tkrn). This package contains the CLEO LINKix Token Ring Link Level.

  NOTE:
  The SDLC Link Level and Token Ring Link Level packages can be loaded and will operate simultaneously. However, the two SDLC packages (coproc and sib) *cannot* be loaded on the same system. The installation procedure prevents these packages from being loaded on the same system.

- SNA Levels — This package can be installed only *after* the link level package.
  - — linkix_sna_128lu, SNA Level (sna128lu). This package provides support for 128 LUs.
- Feature Level 1 — The packages listed below, except for the NetView package (netman), are used in all SNA configurations. The NetView package is used only in an SNA configuration that uses NetView monitoring capabilities.
  - — linkix_3270, Feature Level 1 (linkx3270). This package contains the CLEO LINKix 3270 Feature.
  - — linkix_mgmt, Feature Level 1 (mgmt). This package contains the CLEO LINKix Management Utilities Feature.
  - — linkix_netman, Feature Level 1 (netman). This package contains the CLEO LINKix Netview Package Feature.

- Feature Level 2 — This package can be installed only *after* the Feature Level 1 packages.
  - — linkix_hte, Feature Level 2 (linkxHTE). This package contains the CLEO LINKix HLLAPI TE Feature.
- VIS Packages — These packages work in conjunction with the CLEO software.
  - — INTUITY CONVERSANT VIS V6.0 Synchronous Host Interface Package. This package includes basic voice system host communications software.
  - — INTUITY CONVERSANT VIS V6.0 Enhanced File Transfer. This package provides automatic transferring of files.
  - — INTUITY CONVERSANT VIS V6.0 3270 NetView Alarm Interface. This package provides the capability to send error messages to the host as OGAs. Refer to Chapter 8, "Data Network Connectivity Alarms," for additional information about the NetView Alarming.
- Package Updates

  Updates to the LINKix software may exist. When installing the software, be sure to install the package as well as any updates. Table 7-1 indicates the naming conventions for the update structure. Note that the package name is followed by a two digit number (*XX*). There will only be one update per LINKix package to be loaded at a time.

Table 7-1. LINKix Package Update Structure

Level	Package Name	Update Structure
Link Level	Token Ring	tkrn*XX*
	Coproc	coproc*XX*
	FIFO/SIB	sib*XX*
SNA Level	128 LUS SNA	sna128*XX* or sna*XX*
Feature Level 1	3270 Emulation	l3270s*XX*
	Advanced Netview	netman*XX*
	Management Utilities	mgmt*XX*
Feature Level 2	HLLAPI TE	linkxH*XX*

3270 Configuration Information

After the host is configured properly, you must set the VIS parameters to agree with the host's parameters. The following information details both SDLC and token ring configuration information, including the parameters that need to be set on the host and the VIS in the particular configuration.

SDLC Configuration Information

Host Sysgen Data for SDLC

The host sysgen data is the configuration information about the 3270 link on the host. The values of the following parameters in the host sysgen file are critical for the proper functioning of the 3270 software. Refer to "VIS Data for SDLC Environment" for additional information concerning configuration values.

- DLOGMOD — Should be set to D4C32782 or the system default for the ISM 3278 Model 2 terminal
- DUPLEX — Can be either HALF or FULL. On multidrop lines (that is, when more than one terminal shares the line), use HALF duplex.
- MAXDATA — Determines the maximum path information unit for type2. The MAXDATA parameter should be less than or equal to 265.
- MAXOUT — Determines the maximum number of frames sent before the next polling. Set the MAXOUT parameter to 7.
- NRZ (Non Return to Zero) — Should be noted so that Encoding parameter on the VIS can be configured to match the host setting. Set this parameter to either NRZ (Non Return to Zero) or NRZI (Non Return to Zero Inverted).
- PU ADDR (Physical Unit Address) — Critical for host communication. This value is defined as a hexadecimal (that is, base 16) value. The PU address corresponds to the Poll Address parameter on the VIS.

  NOTE:
  In Version 6.0, the Poll Address *must* be a hexadecimal value. In certain previous releases, the VIS required the Poll Address to be a decimal value.

- PUTYPE — Sets the cluster controller type. Set the PUTYPE parameter to 2.
- SPEED — Can be any standard speed up to 56-Kbps baud that is supported by the attached modem or modem eliminator and the interface card. Make sure that it does not exceed the maximum speed of the modems or modem eliminators being used.
- TYPE — Can be either SWITCHED or LEASED. This value corresponds to the Line Type parameter on the VIS. It must match the setup for the modem or modem eliminator. Refer to the information provided later in this chapter for assistance in operating at speeds over 9600 baud.

VIS Data for an SDLC Environment

The SDLC configuration information is stored in a binary file called **/usr/lib/linkix/com.cfg**. The SDLC Protocol screen fields correspond to the configuration parameters described in Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. Other configuration parameters are available through the LINKix command

comconfig; however, they are rarely used and thus are not included in the following list. Refer to the *LINKix Administration Guide* for additional information on the **comconfig** command and other additional configuration parameters.

NOTE:
It is recommended that you use the screen available in the VIS menu to configure the SDLC environment. *Only* those with in-depth knowledge of SDLC connections should use the **comconfig** command.

- Connection Name — Specifies the SDLC connection. Valid values are SDLC*n*, where *n* is the card number. Default values are SDLC1 and SDLC2.
- Card Number — Indicates the card number to be used for this connection. Valid values are 1 (the default) and 2.
- Line Type — Specifies whether the connection is a leased or dial-up connection. Valid values are leased, switched manual dial, and switched autodial. Refer to the "Modem Configurations" information later in this chapter.
- Note Id to Send — Necessary only for host links that go through dial-up lines. The node identification is derived from the host system parameters.
- Encoding — Specifies the data link's encoding format. This parameter should match the settings in the host sysgen data for NRZ parameter. The valid values for encoding are nrz (Non Return to Zero) or nrzi (Non Return to Zero Inverted.)
- Constant Carrier — Specifies whether the modem supports constant carries. If this parameter is Yes, the voice system will keep the Request to Send (RTS) high.
- Poll Address — Specifies the cluster controller address. This value must be a hexadecimal value.

  NOTE:
  In Version 6.0, the Poll Address *must* be a hexadecimal value. In certain previous releases, the VIS required the Poll Address to be a decimal value.

- LU — Specifies which LUs should be defined as 3278 Model 2 terminals. The list of device numbers can range from 2 to 129. These numbers correspond to the LUs that are defined in the host sysgen data. They do *not* have to be consecutive numbers.

Modem Configurations

Certain line configurations must be present to use SDLC baud rates above 9600. Table 7-2 summarizes the affected configuration parameters.

Table 7-2. Modem Configuration Parameters

Constant Carrier	Line Type	Max. Baud	Comments
Yes	Leased	56 Kbyte	Ideal environment for the highest supported baud rates
No	Leased	56 Kbyte	Typical environment for a multi-drop configuration
No	Switched	9600 Kbyte	Not supported at line speeds above 9600 Kbyte

The configuration parameters Line Type and Constant Carrier must be set to reflect your modem environment. The three possible Line Type values are switched manual, switched autodial, and leased. The leased setting indicates that a line connection will always be present. The switched manual and switch autodial settings specify that one end must dial up the other end to establish for a line connection. Switched manual means a number must be manually dialed to make the connection. Switched autodial means that LINKix dials the number to make the connection if the modem allows it.

NOTE:
Defining the dial string for a switched autodial connection must be done through the LINKix utilities. It cannot be done through the INTUITY CONVERSANT administrative screens.

The two possible Constant Carrier values are No and Yes. The No setting is used in most dial-up environments except when the modem is a V.32, V.22, or a V.42. The No setting must also be specified in multidrop environments. The Yes setting is used in single-drop, dedicated-line environments or when V.22, V.32, or V.42 modems are being used.

Table 7-3 summarizes the Constant Carrier and Line Type configuration parameter settings. The Request to Send (RTS)/Clear to Send (CTS) column indicates the action of the RTS modem signal. Toggled means that RTS is raised and lowered as required; Constant means that RTS is raised during protocol initialization and left high.

Table 7-3. Configuration Parameters

Constant Carrier	Line Type	RTS/CTS	Situation
No	Switched	Toggled	All dial-up modems *except* V.22, V.32 or V.42 which keep DCD constantly high
No	Leased	Toggled	Multidrop environments (not dialup)
Yes	Switched	Toggled	Dial-up environments using V.22, V.32, or V.42 modems which indicate line-connection via DCD
Yes	Leased	Constant	Single-drop, dedicated-line environments

Token Ring Configuration Information

Host Sysgen Data for a Token Ring Environment

The host sysgen data is the configuration information about the 3270 link on the host for a token ring environment. Refer to “VIS Data for Token Ring” for additional information.

- PU ADDR — An entry in this field must exist, but its value is not significant for the VIS.
- MAXDATA — This parameter determines the maximum size path information unit. With the default LINKix configuration, this should not exceed 1929. If LINKix and the UNIX kernel are tuned to allow a larger frame size, MAXDATA can grow to 4105.
- MAXOUT — This parameter determines the maximum number of frames sent before the next polling. Set the MAXOUT parameter to 7.
- PUTYPE — This parameter sets the cluster controller type. Set the PUTYPE parameter to 2.
- IDBLK, IDNUM — These values in combination must correspond to the “Node ID to Send” parameter on the VIS. See “VIS Data for a Token Ring Environment” for additional information on the Node ID to Send parameter.

VIS Data for a Token Ring Environment

NOTE:
It is recommended that you use the screens available in the VIS menu to configure the token ring environment. *Only* those with in-depth knowledge of token ring connections should use the **comconfig** command. Refer to "Adding Token Ring Protocol" in Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

- Connection Name — Specifies the token ring connection. The default connection name is TKR*n,* where *n* is the first unused number starting from 1. (For example, if the current connection names are TKR1 and TKR3, TKR2 is the default.)
- Adapter Device name — Specifies the particular adapter used by this connection. The default is ibmtok_0.
- Local SAP Address — Specifies the Service Access Point (SAP) address the remote nodes use to contact the VIS. This is a 2-digit hexadecimal number ranging from 04 to EC. The value must be a multiple of 4. The default is 4.
- Remote SAP Address — Specifies the remote computer's SAP address. This is a 2-digit hexadecimal number ranging from 04 to EC. The value must be a multiple of 4. The default is 4.
- Node ID to Send — Specifies the ID that is to be sent to the remote computer. This is an 8-digit hexadecimal number ranging from 00100001 to FFEFFFFF.
- Remote Network Address — Specifies the address of the host remote token ring node to which the VIS is connecting. This is a 12-digit hexadecimal number ranging from 000000000000 to FFFFFFFFFFFF. There is no default.
- LU — Specifies which LUs should be defined as 3278 Model 2 terminals. The list of device numbers can range from 2 to 129. These numbers correspond to the LUs that are defined in the host sysgen. They do *not* have to be consecutive numbers.

Administration Interfaces

The 3270 Synchronous Communications software can be administered from either the screen interface or the command line interface. This section details administration of the SDLC and token ring protocols from the command line. Many of the commands listed are also available from the VIS screen interface. For information on host administration via the screen interface, refer to Chapter 1, "User Interface," and Chapter 3, "Configuration Management," of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591.

Using Host Interface Commands

The following host interface commands are used in administering and maintaining the host interface environment and gathering network diagnostic information. Both Lucent Technologies and CLEO Communications have developed commands to support the host interface software on the VIS.

Session Numbering Conventions

Many of the commands described in this section require you to specify the session on which the command is to be performed. VIS commands require host session numbers. LINKix commands require HLLAPI session IDs. The host session numbers range from 0–127. The HLLAPI session IDs range from 2–129 and are equal to the host session number plus 2. The host session numbers are assigned dynamically when the user configures the LUs and stops and restarts the voice system. The mapping from connection name and LU number to host session number is provided on the Display Host Sessions screen as described in Chapter 3, “Configuration Management,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. If only one connection is configured and consecutive LUs are configured starting at 2, the HLLAPI session IDs are equal to the LU numbers.

For example, if SDLC 1 is configured with LUs 2–33 and SDLC 2 is configured with LUs 2–33, host session 0 equals HLLAPI session 2 which also equals LU2 on connection SDLC 1. Also, host session 32 equals HLLAPI session 34 which also equals LU2 on SDLC 2.

VIS Commands

For additional information concerning these commands, refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration 585-310-591*.

- **sb_te [*<session range or session number>*]**

  This terminal emulation program allows a user to step through the host application, including the log-on, log-off, and recovery procedures of a Script Builder application. This session number or range is optional and can be from 0–127. If a session range is used, it can only include 10 sessions. If no session number is given, the command opens all available sessions installed in the system and automatically displays the first session (use CTRL V to display multiple sessions). If a session is not specified, the system assumes the value “all.” The following are examples of valid **sb_te** commands:

  sb_te

  sb_te 5

  sb_te 5-14

NOTE:
Use the Display Host Sessions screen in the VIS menu to provide the mapping of connection name and LU number to the session number.

Use the **sb_te** command to verify if there have been any changes to the host application. Sometimes changes can occur on the host end that are not passed down to the VIS development end. These discrepancies result in error messages being logged on the VIS and the session stays in recovery. The session number chosen must be released from the host interface process before you invoke **sb_te**. To do this for non-Script Builder applications, stop the DIP. To do this for Script Builder applications, use the **hfree** command.

Use the following procedure to start terminal emulation.

1. Turn on the modem or modem eliminator.
2. Start the 3270 Terminal Emulation software directly by entering:

   sb_te *<session_number or session range>*

   The Terminal Emulator (TE) displays the current screen of the LU. The 3270 status line appears at the bottom of the screen to inform you whether the host is active. Refer to Appendix B, "Status Line Information," of the *3270 User's Guide* for information about the indicators shown in the 3270 status line and what those values mean.

   NOTE:
   The status line of the screen will display the HLLAPI session ID. This value equals the host session number plus 2.

3. If you have dial-up connections, connect with the host computer by dialing the telephone number of the host. If you have direct connections (leased-line), the host will probably identify itself soon after the communications card is loaded.

   You can now to send commands to the host.

sb_te executes the HLLAPI TE. (Refer to the *LINKix User's Guide* for additional information.) The INTUITY CONVERSANT System V6.0 host software provides a new look and feel to the TE. Some important keystrokes to remember are:

[CONTROL] [V] – Goes to the next session

[CONTROL] [U] – Displays the LINKix command menu

[CONTROL] [X] – Exits the terminal emulator

[CONTROL] [Z] – Escape to the UNIX prompt

[ESC] [R] – Resets the keyboard

[ESC] [B] – Captures a screen

- **hspy *<session_number or range or all>***

  By specifying a session number (or all), this command shows what screen currently is being presented on that session. Make a note of this information; it will help to isolate what screens might be involved in the problem.

- **hlogin *<host application>*** or **<session_number or range or all>**

  The **hlogin** command invokes the log-in procedure that is defined in the application's host session maintenance section. This command is often used in the system's cron table to log in early the next morning. It is a clean, convenient way to log in to the host application. Note that the session must be in the logged out state before you can use the **hlogin** command.

- **hlogout *<host application>*** or ***<session_number or range or all>***

  The **hlogout** command invokes the log-out procedure that is defined in the application's host session maintenance section. This command is often used in the system's cron table to log off of the host before it goes down at night. It is a clean, convenient way to log out of the host application. Note that the session must be in the logged in state before you can use the **hlogout** command.

- **hfree *<host application>*** or ***<session_number or range or all>***

  The hfree command releases sessions from their Script Builder application assignments. You must use this command to switch from the application to the TE on a given session. Note that the **hfree** command does not automatically log out the specified session.

- **hassign *[hostsvc] <host application>* to *<session_number or range or all> [FTSCRT]***

  The **hassign** command assigns applications to session numbers. It is necessary to use this command to switch from using the terminal emulator to having an application assigned to a given session. Note that the **hassign** command automatically attempts to log in the specified session. Use the optional FTSCRT argument to assign a session for file transfer.

- **hdelete *[hostsvc] <host application>* from *<session_number or range or all>***

  The **hdelete** command invokes the log-out procedure that is defined in the application's host session maintenance section, releases LUs from their Script Builder application assignments, and automatically removes the host application from the session.

- **hnewscript *<host application>***

  The **hnewscript** command updates the system memory with the latest copy of the specified host application. This command is required to place an updated version of the host application into effect.

- **hdisplay *[host application]***

  The **hdisplay** command displays the host applications that have been verified and installed on the system, as well as the current session assignments for each host application.

- **hstatus *<host application>*** or **<session_number or range or all>**

  The **hstatus** command displays the current status of the host application assigned to the associated session numbers. The command is useful when isolating problems with host applications and checking the number of sessions involved on a call.

- **hconfig**

  The **hconfig** command is the command interface you use to view and modify host configuration files. There are a number of options to the **hconfig** command. If any changes are made to the host configuration, you must restart the voice system for those changes to take effect.

- **hdiagnose conn <connection name>**

  The **hdiagnose conn** command determines which SDLC host interface card is installed (PC/XL or FIFO-SIB) and runs the appropriate diagnostic for that card and software. You must stop the voice system to run this command. If you do not stop the host interface process, **hdiagnose** runs **stop_hi** prior to running the diagnostics and then run **start_hi** after diagnostics are complete.

- **hdiagnose info <connection name>**

  The **hdiagnose info** command provides SDLC configuration information. You must stop the voice system to run this command. If you do not stop the host interface process, **hdiagnose** runs **stop_hi** prior to running the diagnostics and then run **start_hi** after diagnostics are complete.

- **start_hi**

  The **start_hi** command starts the host interface software.

  CAUTION:
  *Normally, the **start_hi** command is not run by the user. This command should not be run when the voice system is at run level 4, as this command is run automatically when the system initializes.*

- **stop_hi**

  The **stop_hi** command stops the host interface software.

  CAUTION:
  *Normally, the **stop_hi** command is not run by the user. This command should not be run when the voice system is at run level 4.*

LINKix Commands

For additional information about the following commands, refer to the LINKix documentation that accompanied your software. Specific references to the LINKix documentation for each command are provided.

- **comconfig** — Use this command to define complex configurations not supported through the screen interface provided on the VIS. Refer to Chapter 4, “Configuration Overview,” of the *LINKix Administration Guide*.

  NOTE:
  It is recommended that you use the screen interface described in Chapter 3, “Configuration Management,” of *Intuity*™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, when configuring the host interface.

- **h3270** — Use this command to enter the HLLAPI TE. Invoking the **sb_te** command calls **h3270**. Refer to Appendix D, “HLLAPI Terminal Emulator,” of the *3270 User’s Guide*.

- **comsend** — Use this command to send a file to the host. When using this command, you must be logged in as root and identify the HLLAPI session ID on which the transfer will be performed. Refer to Chapter 4, “Transferring Files,” of the *3270 User’s Guide*. The HLLAPI session ID is equal to the host session number plus 2. You must specify this session ID as a hexadecimal value. For example, host session 10 uses HLLAPI ID 0xC.

  NOTE:
  The host configuration for file transfer must use the write structured fields (WSF), also known as DFT mode, in the logmode table entry. You must verify this entry in both upgrades and new systems as this is a change from previous product releases.

- **comreceive** — Use this command to receive a file from the host. When using this command, you must be logged in as root and identify the HLLAPI session id on which the transfer will be performed. Refer to Chapter 4, “Transferring Files,” of the *3270 User’s Guide*. The HLLAPI session id is equal to the host session number plus 2. You must specify this session ID in as a hexadecimal value. For example, host session 10 uses HLLAPI id 0xC.

  NOTE:
  The host configuration for file transfer must use the write structured fields (WSF), also known as DFT mode, in the logmode table entry. You must verify this entry in both upgrades and new systems as this is a change from previous product releases.

- **comprintcfg** — Use this command to create a printable version of the configuration file. Refer to Chapter 4, “Configuration Overview,” of the *LINKix Administration Guide*.

- **comservice** — Use this command to turn tracing on and off for LINKix processes. This command is used for SDLC or SNA protocol traces. Refer to Chapter 13, “Advanced Diagnostics,” of the *LINKix Administration Guide.*
- **combrowse** — This command is a utility for viewing and filtering trace files.

Administering File Transfer

You can perform file transfer either interactively through the screen interface or directly via UNIX commands. To perform file transfer interactively via the VIS screens, use the File Transfer option provided via the Terminal Emulator selection in the Command Menu. For information on performing file transfer using either method, refer to Chapter 4, “Transferring Files,” of the *3270 User’s Guide*.

Interactive File Transfer

Refer to Chapter 3, “Controlling 3270 Emulation,” of *3270 User’s Guid*e for information on

- Accessing the main screen and navigating through its menus
- Controlling display sessions
- Controlling printer sessions
- Viewing host response times
- Sending NetView alert messages
- Exiting and resuming 3270 emulation

Direct File Transfer

To perform file transfers directly, use the **comsend** and **comreceive** programs in the directory **/usr/bin**. These programs transfer files using a screen-buffer that interacts with the host IND$FILE file transfer program.

NOTE:
You must be logged in to the host session and at the system ready prompt before you can execute the **comsend** and **comreceive** commands. You must use the **sb_te** command to establish the host session before using the file transfer command.

comsend

Use the **comsend** program to transfer a file from the VIS to the host mainframe; that is, to upload a file. Following is an example of the **comsend** program:

comsend -h 0x*N unix_file host_filename options*

- *-h* is an argument indicating the HLLAPI ID number used to send files. *N* is a value for this argument. The value for *N* ranges from 2 through 129 (0x81). You must specify these values as hexadecimal.

- *unix_file* is the name of the VIS file to be transferred. Note that the naming convention of the file follows UNIX standards. The file must be named with a full path. No directory is required if the file is in the current working directory. Refer to Table 7-4 for tips on how to specify filenames when performing file transfers.
- *host_filename* is the name of the target host mainframe file.
- You can enter several *options* to control the file transfer. These options are provided in Chapter 4, “Transferring Files,” of the *3270 User’s Guide*. Note that some options are not available with all systems.

  NOTE:
  Mainframe systems vary in their requirements for the options list. Some mainframes require that the option list be enclosed in parentheses, some require only the left parentheses, and others do not permit the use of parentheses. You should therefore verify the requirements of the mainframe you are using before using any of these options. All meta characters [* , ( ) etc.] must be preceded by a backslash (\) character in the **comsend** command line. Other characters may work, but the backslash is recommended in all cases.

comreceive

Use the **comreceive** program to transfer a file from the host mainframe to the VIS; that is, to download a file. Following is an example of the **comreceive** program:

comreceive -h 0x*N unix_file host_filename options*

- *-h* is an argument indicating the HLLAPI ID number used to receive files. *N* is a value for this argument. The value for *N* ranges from 2 through 129. You must specify this value as hexadecimal.
- *unix_file* is the name of the target VIS file on download. Note that the naming convention of the file follows UNIX System standards. The file must be named with a full path. No directory is required if the file is in the current working directory. Refer to Table 7-4 for tips on how to specify filenames when performing file transfers.
- *host_filename* is the name of the host mainframe file to be transferred.
- You can enter several *options* to control the file transfer. File transfer options are provided in Chapter 4, “Transferring Files,” of the *3270 User’s Guide*. Note that some options are not available with all systems and may not all be available with all systems.

  NOTE:
  Mainframe systems vary in their requirements for the options list. Some mainframes require that the option list be enclosed in parentheses, some require only the left parentheses, and others do not permit the use of parentheses. You should therefore verify the

requirements of the mainframe you are using before using any of these options. All meta characters [* , ( ) etc.] must be preceded by a backslash (\) character in the **comreceive** command line. Other characters may work, but the backslash is recommended in all cases.

When an ASCII file is received from the host, it may have been sent with a ^Z (CONTROL Z) at the end of the file. When you try to "vi" the file, a message may complain about an unrecognized character. You should attempt to get rid of the character in the file. This is typically a problem with TSO and VM systems.

When a binary file is received from the host, zeros (0) are added to the end of the block to make it a multiple of 80. For example, if a file of 4 bytes is sent to the host, it may contain 76 more bytes when it is returned (4 + 76 = 80).

Table 7-4. Filename Guidelines for File Transfer

If Filename Contains	UNIX			Host3270		
	Syntax	Examples		Syntax	Examples	
		Original	Converted		Original	Converted
| & ; < > () ‘ \ ‘ * ? [] # ~ †	Precede each special character with a backslash (\)	ix’yy’a\bc	x\’yy’\a\\bc	Precede each special character with a backslash (\)	#AB~C*D E?cde#f*h	\#AB~C*D E\?cde#f*h
dollar sign ($)	Precede $ with backslash (\)	AB$tmp	AB\$tmp	Precede $ with backslash (\) ‡	XXyy$zz	XXyy\$zz
at sign (@)	Precede @ with backslash (\)	AB@tmp	AB\@tmp	Precede @ with backslash (\) §	XXyy@zz	XXyy\@zz
period (.)	No special syntax	s.xx.c	s.xx.c	Enclose filename first with a backslash (\) followed by an apostro-phe (‘) \’ ††	s.xx.c	\’.xx.c\’
Any char-acter not shown above	No special syntax	abcd	abcd	No special syntax	a123bcd	a123bcd

† Protect # and ~ with a backslash only if they begin the filename

‡ Protect $ with backslash only when the file transfer is done directly with the **comsend** or **comreceive** commands. Do not protect $ when file transfer is done through the 3270 terminal emulator.

§ Protect @ with backslash only when the file transfer is done directly with the **hsend**, **comsend** or **comreceive** commands. Do not protect @ when file transfer is done through the 3270 terminal emulator.

†† Protect . only if transferring files to/from a tso system and the dots in the filename are a fully qualified filename (containing the user id).

§§ You may *not* use an underscore when specifying a filename.

Administering Enhanced File Transfer

Local VIS Procedures

The user at the local VIS should do the following:

1. Develop, verify, and install a host maintenance script that initiates and maintains a host session; that is, provides procedures for login, logout, and recovery screen sequences. Note that the script should leave the host session at the host system ready prompt that will allow an interface with the host IND$FILE file transfer program. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760, for information on developing, verifying, and installing a host maintenance script.

   NOTE:
   After a file transfer, the host system-ready prompt may be in a different position on the screen. The recovery and logout sequences must take this into consideration. The user may need to define multiple screens for the host system-ready prompt.

2. Begin the file transfer by executing the **hassign** command to assign the host maintenance script to the host session. Following is the format of the **hassign** command:

   hassign <*application*> to <*session*> FTSCRT

   application is a required argument that specifies the host maintenance script for file transfer. *session* is a required argument that specifies the session number or a range of session numbers. *FTSCRT* is a required argument that assigns the session for file transfer. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for information on using the **hassign** command.

3. Execute the **hstatus** command to verify that the session is logged in to the proper screen for file transfer. If the session is logged in properly, **hstatus** displays “file transfer” as the session’s status. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for additional information on using the **hstatus** command.

4. If you are preparing to transfer a Script Builder application script to the remote site via the host, you must develop, verify, and install this application script using Script Builder. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760, for information on developing, verifying, and installing Script Builder applications.

5. If you are preparing to transfer a Script Builder application script to the remote site via the host, create a batch file to remove existing applications and install the new application script developed in the previous step. This batch file is sent with the application script to the remote VIS via the host. Once the remote VIS receives the batch file, it executes the commands in the batch file. The batch file can be any combination of regular UNIX

commands, executable shell files, and executable program names. For example, to automatically install an application received from the host, the batch file can execute the **remove_appl**, **restore_appl**, and **install_appl** commands. Note that the name of the batch file should end with **.vb**. Procedures and suggestions for creating batch files are described in detail later in this chapter under "Batch Files used in the Enhanced File Transfer System." Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for information on the **remove_appl**, **restore_appl**, and **install_appl** commands.

6. If you are preparing to transfer a Script Builder application, execute the **backup_appl** command to create one file each for the transaction, speech, and database portion of the transaction. Next, bundle the Script Builder transaction, speech, and database files and the batch file into one bundle using the UNIX **cpio** command. If you are preparing to transfer a software package, bundle the software package and the batch file into one bundle by using the **cpio** command. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for information on the **backup_appl** command. Refer to the *UNIX System V Release 4.2 Command Reference (a–l)* for information on using the UNIX **cpio** command.

7. Name the file to be sent to the remote VISs and if necessary modify the DESTINATION parameter in the configuration file (**/vs/data/fts_config**) on the local VIS machine to include this filename. The DESTINATION parameter specifies the name of the bundle on the host 3270 mainframe. The DESTINATION parameter is required and must be set either in the configuration file or on the **hsend** command line. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for information on using the **hsend** command.

8. Send the file to the host by executing the **hsend** command. The format of the **hsend** command is as follows:

   hsend file=<filename> ***[dest=] [opt=]***

   <filename> is a required argument that specifies the full path name of the UNIX file or cpio bundle to be sent to the host. Refer to Figure 7-4 for filename guidelines for file transfers. *dest* is an optional argument that specifies the final destination of the file at the host. If a destination is not specified, the DESTINATION parameter from the **/vs/data/fts_config** file is used as the destination. *opt* is an optional argument that specifies either a list of options or the letter n (for no options). Note that the options must be separated by a space. If an option list is provided, it is sent to the host. If the option argument value is *n*, the PARAM1, PARAM2, and PARAM3 parameter values are not appended to the host IND$FILE file transfer program. If this argument is missing, the PARAM1, PARAM2, and PARAM3 parameter values are used.

The local VIS is now ready to send files to the remote VIS via the host and/or receive files sent from the remote VIS via the host.The procedures for sending

files from the host to the remote VIS and sending files from the host to the local VIS are discussed later in this chapter.

Remote VIS Procedures

The user at the remote VIS should do the following:

1. Develop, verify, and install a host maintenance script that initiates and maintains a host session; that is, provides procedures for login, logout, and recovery screen sequences. Note that the script should leave the host session at the host system ready prompt that will allow an interface with the host IND$FILE file transfer program. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760, for information on developing, verifying, and installing a host maintenance script.

   NOTE:
   After a file transfer, the host system-ready prompt may be in a different position on the screen. The recovery and logout sequences must take this into consideration. The user may need to define multiple screens for the host system-ready prompt.

2. Execute the **hassign** command to assign the host maintenance script to the host session. Following is the format of the **hassign** command:

   hassign *<application>* to *<session>* FTSCRT

   application is a required argument that specifies the host maintenance script for file transfer. *session* is a required argument that specifies the session number or a range of session numbers. *FTSCRT* is a required argument that assigns the session for file transfer. Refer to *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for information on using the **hassign** command.

3. Execute the **hstatus** command to verify that the session is logged in to the proper screen for file transfer. If the session is logged in properly, **hstatus** displays “file transfer” as the session’s status. Refer to *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for additional information on using the **hstatus** command.

4. Modify the **/vs/data/fts_config** configuration file on the remote VIS to poll the local VIS for the file. Procedures for modifying the **/vs/data/fts_config** file are described later in this chapter.

5. If necessary, create the APPL_FTS utility to preprocess the bundle received from the host. Preprocessing is optional and may be used to customize the file transfer feature by adding header information, special files, etc. to the files that will be handled by the Enhanced File Transfer system. Note that the full path name of the preprocessing file should be added to the APPL_FTS field in the configuration file.

The remote VIS is now ready to receive files sent from the local VIS to the host and/or send files to the host and the local VIS. The procedures for sending files

from the host to the local VIS and from the host to the remote VIS are discussed later in this chapter.

Receiving Files from the Host on the Remote VIS

The Enhanced File Transfer System automatically transfers files from the host to the remote VIS. This procedure is part of this automatic transfer:

1. Poll the host at a time determined by the **/vs/data/fts_config** configuration file (POLL_START, POLL_FREQ, and POLL_END fields).
2. Receive a bundle sent by the local VIS.
3. Place the bundle in a temporary directory (for example, fts_tmp1) under the directory specified in the FROM_HOST_DIR field in the **/vs/data/fts_config** file. By default, each temporary directory is created under the **/usr/tmp** default directory.
4. Create a log file with the full path name of the bundle as specified in the FROM_HOST_DIR field in the **/vs/data/fts_config** file. All batch file command outputs are appended to the log file, with each record in the log file containing the original command line and the command output.
5. Execute either the APPL_FTS file (if one exists) or the UNIX **cpio** command (if an APPL_FTS file does not exist) to preprocess the bundle received from the host.
6. After preprocessing is complete, execute the batch file received from the host under the temporary directory. Note that the batch file name must end with .vb and must conform to UNIX standards.

   WARNING:
   If more than one batch file is sent in a bundle, the transfer is treated as an error and no further processing will take place for that bundle.

7. Record the status of all Enhanced File Transfer activities in the log file.
8. After executing the batch file, the Enhanced File Transfer system sends the log file to the host. At this time, the user may execute the **hsend** command to send other files to the host, including output files created during the execution of commands within the batch files. Refer to the information on “Sending Files to the Host” later in this chapter for information on this procedure.
9. Set the next polling time.

Sending Files from the Remote VIS to the Host

Use the **hsend** command to send files other than the log file from the remote VIS to the host. These files can include output files created during the execution of commands within the batch files. The format of the hsend command is as follows:

hsend file=<*filename*> *[dest=][opt=]*

<filename> is a required argument that specifies the full pathname of the UNIX file or cpio bundle to be sent to the host. Refer to Table 7-4 for filename guidelines for file transfers. *dest* is an optional argument that specifies the final destination of the file at the host. If you do not specify a destination, the DESTINATION parameter from the **/vs/data/fts_config** file is used. *opt* is an optional argument that specifies either a list of options or the letter "n" (for no options). Note that you must separate the options by a space. If an option list is provided, it is sent to the host. If the option argument value is *n*, the PARAM1, PARAM2, and PARAM3 parameter values are not appended to the host IND$FILE file transfer program. If this argument is missing, the PARAM1, PARAM2, and PARAM3 parameter values are used.

NOTE:
The Enhanced File Transfer system removes the log file on the remote VIS after the file is successfully transferred to the host. If the log file is not sent to the host successfully, it is stored at the FROM_HOST_DIR directory and renamed ***[unix time]*.log** where *[unix time]* indicates the current system time in seconds. It is the user's responsibility to remove the stored log file later to save disk space.

Batch Files used in the Enhanced File Transfer System

UNIX commands have two output files, stdout and stderr. Conventionally, stdout is used for expected output (often none) and stderr is used for error messages. You can discard the output of either the stdout, stderr, or both by directing it to **/dev/null**. Generally, a command line in a batch file behaves the same way as a command typed at a terminal; that is, the following occurs:

- Undirected stderr and stdout are collected and appended to the host log
- If stdout is redirected to **/dev/null**, the output is not appended to the host log (for example, **install_sw xmas_sale > /dev/null**)
- If stderr is redirected to **/dev/null**, the output is not appended to the host log (for example, **install_sw xmas_sale 2 > /dev/null**)
- If both stderr and stdout are redirected to **/dev/null**, nothing regarding the command is written to the host log (for example, **install_sw xmas_sale > /dev/null 2 > &1**)

All batch file command outputs are appended to the log file that is created in the FROM_HOST_DIR. Each record in the log file contains the original command line and the command output.

NOTE:
Do not include commands that are inherently interactive or that do not terminate automatically in batch files. Commands that are inherently interactive are difficult to execute on a noninteractive basis unless all the required responses are known in advance. Commands that do not terminate automatically can also cause a problem.

Configuring fts_config File for Enhanced File Transfer

The Enhanced File Transfer configuration file contains field settings that are used in configuring the IND$FILE file transfer program on the host.

Configuration information is stored in an ASCII file called **/vs/data/fts_config**. Use the following procedure to view and edit the contents of this file:

1. Log in as **root**.
2. Enter **vi /vs/data/fts_config**

   The default value for parameters in **/vs/data/fts_config** are shown in Figure 7-5.

```
POLL_START=-01:00
POLL_FREQ=04:00
POLL_END=24:00
DESTINATION=
ORIGINATION=
APPL_FTS=
HOST_OS=TSO
FROM_HOST_DIR=/usr/tmp
PARAM1=
PARAM2=
PARAM3=
Verbose=1
Max_receive=1
```

Figure 7-5. /vs/data/fts_config Example

Following is a description of each field in the **/vs/data/fts_config** file.

POLL_START

The POLL_START field specifies the time of day at which the Enhanced File Transfer System first polls the host. The default value is -01:00. This specifies that the Enhanced File Transfer System never polls the host but sends files only when a request is made. If you change the POLL_START value from the default (-01:00) to any value between 00:00 to 24:00, the Enhanced File Transfer System uses the new POLL_START value following the next polling period or the next **hsend** command.

NOTE:
You may not set the POLL_START field to a value greater than 24 hours (24:00). If you attempt to set the POLL_START field to a value greater than 24 hours, the value (00:00) is used.

POLL_FREQ

The POLL_FREQ field specifies the intervals at which the Enhanced File Transfer System polls the host. The default value is 04:00. This specifies that polling will occur every 4 hours. If you set the POLL_FREQ field to a value less than or equal to 00:00, the Enhanced File Transfer System polls only at POLL_START. For example, if the POLL_FREQ field is set to -01:00 and the POLL_START is set to 01:00, the Enhanced File Transfer System polls the host starting at 01:00. If you set the POLL_FREQ field to a value greater than 24 hours, the Enhanced File Transfer System polls the host at this offset from POLL_START. For example, if you set the POLL_START to 02:30 and POLL_FREQ to 50 hours, the Enhanced File Transfer System polls the host at 4:30 a.m. on alternate days. If you change the POLL_FREQ field just after the most recent POLL_START, the Enhanced File Transfer System changes the POLL_FREQ at the next POLL_START or the next execution of the **hsend** command. For example, if POLL_FREQ is changed from 01:00 to 00:30 at 2:20 p.m., the POLL_FREQ does not change until the next polling period begins at 3:00 p.m. or until the **hsend** command is executed.

POLL_END

The POLL_END field indicates the time of day after which the Enhanced File Transfer System will not poll the host. The default value is 24:00.

NOTE:
You may not set the POLL_END field to a value less than or equal to 00:00 or greater than or equal to 24:00. If you attempt to set POLL_END in this manner, the default value (-01:00) is used. This default value indicates that the POLL_END field should be ignored. Only the POLL_START field is used to determine whether to begin polling.

DESTINATION

The DESTINATION is a required field that specifies a dataset (file) name that is acceptable to the host. The DESTINATION specified in this field is used as the destination argument to the **hsend** command for sending a bundle to the host.

ORIGINATION

The ORIGINATION is a required field that indicates a dataset (file) name that is acceptable to the host. The ORIGINATION specified in this field is used as the origination argument to the **comreceive** command for receiving a bundle from the host.

APPL_FTS

The APPL_FTS field is used only if a program has been created to preprocess the bundle received from the host. The APPL_FTS field specifies the full path name of this program. The APPL_FTS default value is NULL. This default value indicates that a preprocessing program does not exist.

HOST_OS

The HOST_OS is a required field that indicates the name of a host application. You may specify either CICS, TSO, or CMS in this field. The HOST_OS default value is TSO.

FROM_HOST_DIR

The FROM_HOST_DIR field specifies the full pathname of the directory on the VIS where the Enhanced File Transfer System creates a temporary directory to receive a bundle from the host and executes the batch file from each of these temporary directories. The FROM_HOST_DIR default value is **/usr/tmp**.

PARAM1, PARAM2, PARAM3

PARAM1, PARAM2, PARAM3 are optional fields that are reserved for any additional parameters. Note that the parameters are sent in order of PARAM1, PARAM2, and PARAM3 with a space in between them (for example, PARAM1 PARAM3). Refer to Chapter 4, "Transferring Files," of the *3270 User's Guide* for a list of file transfer options.

Verbose

The Verbose field indicates the level of detail of the **/tmp/fts_trace** file. A Verbose setting of 1 (the default) indicates the most detailed level. This file is used for debugging purposes. A Verbose setting of -1 instructs the VIS to turn off tracing.

Max_receive

The Max_receive field specifies how many times the VIS attempts to receive a bundle from the host during each polling cycle. The Max_receive default value is 1. A Max_receive value of -1 specifies that the VIS will never poll the host.

Changes in the configuration file take effect the next time the host is polled. To make the changes take effect immediately, perform the "Stopping the Voice System" and "Starting the Voice System" procedures in the appropriate Intuity ™ CONVERSANT® System Version 6.0 maintenance book. You can also cause changes to take effect by using the **hsend** command. Refer to the information on sending files to the host in this chapter for additional information on using the **hsend** command.

Examples of Enhanced File Transfer

Sending a Single ASCII File to the Host

Enter:

hsend file=<*filename*> *[dest=filename on the host] [opt=ASCII CRLF]*

NOTE:
The above example assumes that PARAM1 and PARAM2 are set to "ASCII" and "CRLF" and DESTINATION is set to the host dataset name. If these values are not set, the *<dest>* and *<opt>* fields are not optional.

Receiving a Single ASCII File from the Host

1. Make sure that polling is on.
2. Create the file **/usr/tmp/appl** with the following contents, where */usr/tmp/hostfile* is the file received from the host:

   cp /usr/tmp/fts_tmp1/tmp1.pkg /usr/tmp/hostfile
3. Enter **vi /vs/data/fts_config**
   a. Change the APPL_FTS parameter to **/usr/tmp/appl.**
   b. Change the FROM_HOST_DIR parameter to **/usr/tmp.**
   c. Change the PARAM1 parameter to ASCII and the PARAM2 parameter to CRLF.
   d. Change the ORIGINATION parameter to the filename on the host.

Receiving a Package from the Host

Make sure that polling is on and modify the **/vs/data/fts_config file** as follows:

1. Keep the APPL_FTS parameter blank.
2. Change the FROM_HOST_DIR parameter to **/usr/tmp.**
3. Change the ORIGINATION parameter to the destination file name used in the **hsend** command.
4. Change the DESTINATION parameter to a desired host file name for later use. The Enhanced File Transfer System will use this file name in sending the trace log from the **tmp.vb** execution back to the host.

Sending/Receiving an Application

The following demonstrates the steps necessary to test sending an application to a host, and then receiving that same application back through the use of Enhanced File Transfer:

1. Enter **backup_appl -n *<appl_name>***

   This creates binary files for each component of an application, which include Transaction (Trans), Speech (Spch), and Database (Dbase).
2. Enter **cd /tmp/sb/BkUpAppl/*<appl_name>***

This is the directory to which the Trans, Spch, and Dbase files are copied.

3. Enter **vi *<filename>*.vb**

   This is the file that will be run when it is received on the target system.

4. Enter **ls |cpio -oBcv > *<all_files_name>***

   This creates the <*all_files_name*> bundle that contains all the files together and will be sent using **hsend.**

5. Enter **vi /vs/data/fts_config**

   a. Update the Destination parameter with the name you want this application to be stored under on the host system. Remember, it must conform to the host file naming rules and special characters should be preceded with a backslash.

   b. Update the POLL_START with a positive value that you want to use to poll the host.

   c. Make sure that PARAM1, PARAM2, and PARAM3 are set to blank.

6. Enter **hassign *<eft_appl>* to *<session number>* FTSCRT**

   This assigns the Enhanced File Transfer script to a session and gets a session to the READY prompt, ready for a file transfer. To ensure that the session is ready, enter **hstatus *<session number>***. This session number must specify the the “file transfer” state.

7. Enter **hsend file=/tmp/sb/BkUpAppl/*<appl_name>*/*<all_files_name>***

   This starts the send of the <*all_files_name*> to the host, using the session hassigned in Step 6.

8. Enter **vi /vs/data/fts_config**

   Set DESTINATION to blank and set ORIGINATION to the name you stored the application under on the host in Step 5. Once the send has completed, this file is updated when the polling value is reached, and the receive command is initiated. Once the file is received the ***<name>*.vb** file is run. Some examples of what might be used in the ***<name>*.vb** file are **backup_appl**, **restore_appl**, and/or **install_appl**, to first make a backup of the original application, then to restore the new application, and to finally install the new application. Once the receive is complete, the dates on the appl files in **/att/trans/sb/*<appl>*** should be close to the current time.

Host DIP Parameter File

The host DIP parameter file **/vs/etc/default/agdip3270** allows access to certain parameters that may be useful when designing your host application.

SESSIONS_TO_START Parameter

The SESSIONS_TO_START parameter allows you to specify the number of sessions to which you want to receive and send screens concurrently. Setting this parameter to 5, for example, means that 5 sessions at most are allowed to start logging in, logging out, or recovering at one time. The rest of the sessions wait to start until 1 or more of the 5 sessions complete executing their log-in, log-out, or recover sequences. The default is to allow all 32 sessions to access the host concurrently.

In most cases, the default works well. However, if all 32 sessions are logging-in, an individual session takes longer to log in than it would if it was the only one accessing the host. This is because an individual session has to compete for the host link resource with 31 other sessions.

On the other hand, setting SESSIONS_TO _START=1 allows only one session to log in at a time while the rest wait their turn. This speeds up the logging in for one session, but overall it takes longer to log in all sessions than if multiple session were logging in at a time.

Selecting a suitable value for SESSIONS_TO _START depends on the host environment and the applications and involves some trial and error. However, in most cases the default of 32 is acceptable.

To set the SESSIONS_TO_START parameter:

1. Perform the “Stopping the Voice System” procedure. See the appropriate Intuity ™ CONVERSANT® System Version 6.0 maintenance book, for details.
2. Enter **vi /vs/etc/default/agdip3270**
3. Set the SESSIONS_TO_START parameter to the maximum number of sessions you want to be receiving and sending screens concurrently. For example, to have only one session interacting with the host, set SESSIONS_TO_START=1.
4. Exit the file.
5. Perform the “Starting the Voice System” procedure in the appropriate Intuity ™ CONVERSANT® System Version 6.0 maintenance book.

LOGOFF_TIMEOUT Parameter

The LOGOFF_TIMEOUT parameter specifies the maximum amount of time the **stop_vs** command waits for any active session to be logged out before the voice system is stopped. The default value for LOGOFF_TIMEOUT is 60 seconds. You should increase this value only if **stop_vs** does not allow enough time for all LUs to be logged off. This may be necessary if your system has many LUs or the LUs have lengthy logout sequences.

MAX_NUMBER_OF_LUs Parameter

The MAX_NUMBER_OF_LUs parameter specifies the maximum number of LUs that can be configured for a system. The default value is 128 LUs.

NOTE:
Do not change this value.

AUTORESET_LUs Parameter

The AUTORESET_LUs parameter specifies that the hostdip automatically sends a reset key if the LU is in recovery and input is inhibited. It also sends the system reset key if the LU is in recovery and the screen is the system services control point (SSCP) or UNOWNED. The default value for AUTORESET_LUs is No. This parameter should only be set to Yes if the LUs get stuck in recovery for one of the reasons listed previously in this description.

Retry Strategy

Sessions that repeatedly fail to log in are subject to a retry delay before trying to recover again. The retry delay is incremented by 20 seconds for each consecutive failure. For example, six consecutive failed attempts results in 120 seconds of delay before the session is allowed to start its seventh attempt to log in. The session will wait no longer than 600 seconds to attempt to log in again.

A session is *not* delayed the next time it tries to log in if one of the following occurs:

- The session is freed via **hfree**. This clears all past failed attempts made to log the session in.
- The **hlogout**, **hassign**, **hnewscript**, or **hdelete** commands are executed on the session. These commands are queued if the session is in the middle of executing its log-in or recover sequence. Once the log-in or recover sequences completes, the commands(s) are executed.
- The session recovers and becomes logged-in.

Figure 7-6 shows how a session tries to log in. After a session is assigned a Script Builder application, it begins to log in. After it completes the log-in sequence, the session is in one of the following states:

- The session is **logged in** if the current screen is the transaction base screen. In this state, the session is ready to get data when a call is made to a Script Builder application.
- The session is **logging in** if the current screen is the log-in base screen. In this state, the session will wait an additional 20 seconds before attempting to reach the transaction base screen.
- The session is **recovering**. In this state, the session waits an additional 20 seconds before attempting to reach the transaction base screen.

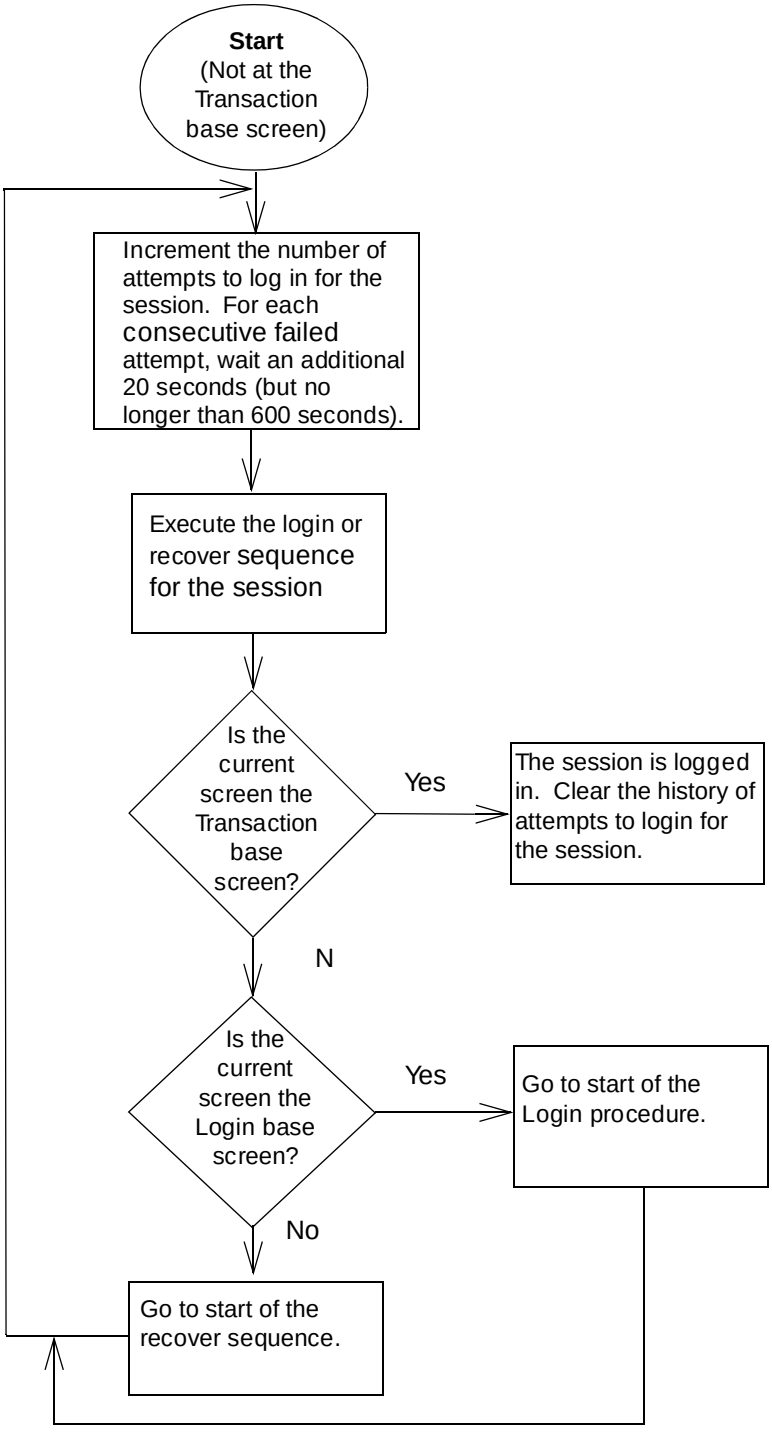


Figure 7-6. How a Session Tries to Log in

Application Development Issues

The following are current application development issues concerning the host interface software.

Intermediate Screens

It has always been important for host applications to deal with intermediate screens. An intermediate screen occurs when the host responds with a screen and unlocks the keyboard (the sign that the voice system can send another screen to the host), but in fact the host is sending another screen. This behavior occurs most frequently during the log-in process.

Because token ring networks are faster than SDLC connections, it is possible that a host application will experience more intermediate screens over a token ring network. If an application is moved from an SDLC environment to a token ring environment, and the log-in sequence does not work as it used to, it is likely that the application is receiving these intermediate screens. If you experience this problem:

- Add recognition criteria to the screen definition to differentiate the intermediate screen from the final screen.
- Add an additional Get Host Screen action between the Send Host Screen Action and the real Get Host Screen action. In the new Get Host Screen action, wait for a screen that will not be sent. This forces a pause in the sequence. Then, the next Get Host Screen executes after the host has had a chance to send all screens.

TCP/IP Communications

Transmission Control Protocol/Internet Protocol (TCP/IP) is a process-to-process protocol. The IP component dispatches information around the network, and the TCP component assures that information's accuracy. TCP/IP within the VIS provides high-speed data transmission over an Ethernet or token ring network.

There are three areas that you must be address when using TCP/IP protocol with the VIS.

- Current network topology — Refer to "Network Architecture" below.
- Application structure — Refer to "Application Development Issues" below.
- Software installation — Refer to "Installing the Optional Feature Software," in the maintenance book that applies to your platform.

Refer to *NFS/RPC/NIS Administration* and *TCP/IP Administration* for additional information on TPC/IP protocol. Refer to the *SQL*NET TCP/IP User's Guide* for additional information on using SQL*NET TCP/IP.

Network Architecture

UnixWare 1.1 includes an implementation of the TCP/IP protocol. The package has been internetworked successfully by AT&T and others with a wide variety of TCP/IP networks. Given this standard and compliant implementation, there is no reason that a VIS running this software cannot be connected successfully to a standard, compliant TCP/IP network.

Figure 7-7 shows the layering of TCP/IP over Ethernet and token ring in the context of the first four layers of the OSI Reference Model. This figure illustrates that the styles of networking differ at the physical and link layer only (Ethernet versus token ring). The network layer and above are the same, regardless of the physical and link layer.

Some standard networking utilities are available with UnixWare. These utilities are used to network the VIS with other machines without developing a custom application interface. These utilities include:

- **rcp** — Allows a user to copy files to and from a remote machine
- **rlogin** — Allows a user to log in to a remote machine from a local machine
- **ftp** — Transfers files to and from a remote network
- **telnet** — Enables terminal and terminal-oriented processes to communicate on a TCP/IP network

Refer to *UNIX SVR4.2 Network Administration* for additional information about standard networking utilities.

Sockets, TLI, and RPC are alternative and equivalent application programming interfaces to the network. Sockets was introduced as part of the UNIX systems 4.2BSD. Almost every implementation of TCP/IP for UNIX includes a sockets interface. TLI was released with AT&T UNIX R3. It offers a streams-based interface to the transport layer. As a streams interface, it offers a measure of portability from one protocol suite to another. RPC is a remote procedure call interface. This implementation of TCP/IP offers a Sockets, a TLI, and an RPC interface.

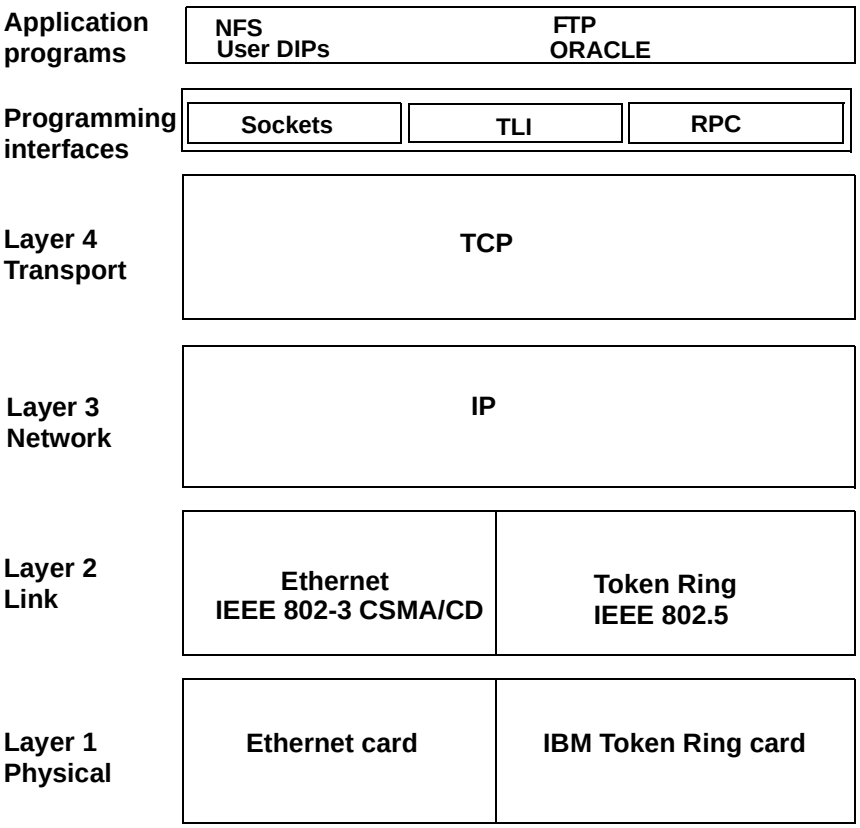


Figure 7-7. Network Layering

Application Development Issues

Typically, a VIS is added to a network that is already in place. Adding a VIS to your network allows you to use information from the network in a custom application. You must first determine if the information you want is available through the standard UnixWare utilities (for example, **rcp**, **rlogin**, **ftp**) or whether a custom process is necessary. Refer to *UNIX SVR4.2 Network Administration* for additional information about the standard network utilities.

If it is necessary to write a custom program, you may also write a data interface process (DIP) to access the program. Refer to *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods* 585-310-761. When writing the DIP, you must use the Sockets, TLI, or RPC application programming interface (refer to *NFS/RPC/NIS Administration*). Within Script Builder you must create an external action to call the **dbase** script instruction to execute the DIP. Refer to “Using Advanced Features,” of *Intuity* ™

CONVERSANT® System Version 6.0 *Application Development with Script Builder* 585-310-760.

It is also possible to use sockets, TLI, or RPC with an INTUITY Response Application Programming Interface (IRAPI) application. Care must be used to determine who the process should block. Refer to chapter 6 of *Intuity* ™ CONVERSANT® System Version 6.0 *Application Development with Advanced Methods* 585-310-761, for information.

Provisioning TCP/IP

The following sections detail the network addressing and hardware and software requirements for the TCP/IP protocol.

Network Addressing

TCP/IP allows each machine on the network to be "addressed" so that it can be distinguished from other machines. Every host on the network must have a unique network address. The addresses consist of four decimal integers each separated by a dot (.). Three different classes of addresses are possible with the TCP/IP protocol. The default network uses a class A address. However, if you want to assume responsibility for maintaining the network database files, other network architectures are possible.

Refer to Chapter 1, "Administering TCP/IP Networks," of *TCP/IP Administration* in the *Novell UnixWare Documentation Set*, 585-310-908, for additional information on setting up the network.

Hardware Requirements

Using the TCP/IP protocol on the VIS requires either an Ethernet or Token/Ring circuit card, depending on the physical and link layer.

The SMC Ethernet card supports the following physical interfaces to the network:

- External transreceiver
- 10BASE2 (ThinNet)
- 10BASET (Twisted Pair)

The IBM token ring card supports the following physical interfaces to the network:

- IBM token ring Network PC Adapter cable
- Category 3, 4, or 5 cable

Refer to "Administering TCP/IP over Ethernet and Token Ring LANs." on page 7-40 for information on installing these network cards.

Software Requirements

The following software packages must be installed on the VIS to use TCP/IP protocol:

- UnixWare 1.1
- The driver specific to the circuit card installed on the VIS (either Ethernet or token ring)

Refer to the next section for installation procedures.

Administering TCP/IP over Ethernet and Token Ring LANs.

This text provides particulars about TCP/IP administration on CONVERSANT systems.

The SMC Ethernet card replaces the StarLAN and InterLAN cards formerly supported. The SMC Ethernet card provides TCP/IP connectivity over an ethernet network. The IBM Token Ring card is also capable of providing TCP/IP connectivity on a token ring network. Because the CONVERSANT system supports both types of TCP/IP interface cards, the installation and administration of TCP/IP can get complicated if both cards are installed in the same system. For that reason this text is divided into several subsections. First are the simpler cases; administration of TCP/IP on systems that contain one or the other interface cards, but not both. The final cases will address setting up TCP/IP when both the SMC Ethernet card and the IBM Token Ring card are installed in a particular CONVERSANT system.

TCP/IP Without Token Ring Card

1. Install the SMC Ethernet LAN card using the instructions found in the appropriate System Installation book. If cabling to the network is available, you may also connect the Ethernet card to the network hub at this time.
2. Refer to the section covering LAN software package installation in the applicable CONVERSANT V6 Maintenance book to begin the software install. There you will make sure that all of the necessary UnixWare TCP/IP packages are loaded on the system. If they are not, follow the instructions to load TCP/IP from the UnixWare tape.
3. Complete the installation procedure found in the applicable CONVERSANT V6 Maintenance book entitled “Installing the EtherCard Driver Package”.

NOTE:
The user responses provided in this section, for instance the IRQ, the I/O case address, the RAM address, the network connection (BNC, Twisted Pair), are simply recommendations. If your situation

calls for a different IRQ or a BNC connection rather than twisted pair, then respond to the prompts accordingly.

4. Following the software installation procedure, refer to the Novell UnixWare guide entitled “TCP/IP & NFS; TCP/IP Administration”. In chapter 3, pages 44-47, perform the following steps:

   Enter: **/etc/confnet.d/configure -i**

5. The configure command will prompt for the interface type. Select “sme_0”.
6. The configure command will prompt for the host name. This is what is known as the network node name for the system, the name of your CONVERSANT system as you want it to appear in the /etc/hosts file. The typical response is to enter the system UNIX name or some alias name for the system.
7. The configure command will prompt for the ifconfig options. (The choices here are: yes, no, ClassC, BerkeleyC, info.) Select “ClassC”
8. From page 47 in the UnixWare TCP/IP Administration guide, change the route daemon to the active mode by editing the /etc/inet/rc.inet file and changing the line that looks like:

   ```
   /usr/sbin/in.routed -q
   ```

   to look like:

   ```
   /usr/sbin/in.routed
   ```

9. Enter: **stop_vs**
10. Run **smc_setup** again to reaffirm the entries that you made during the EtherCard Driver Package installation.
11. Enter: **/etc/conf/bin/idbuild**
12. Then enter: **shutdown -y -g0 -i6**
13. Edit the /etc/hosts file to enter the remaining network node names for your LAN.
14. Run the following command to display the network routing table:

    netstat -r

    The response should look something like this:

```
Routing tables
Destination  Gateway     Flags  Refs  Use  Interface
localhost    localhost   UH     0     0    lo0
xxx.yy.zzz   host_name   U      ?     ?    sme0
```

15. If your SMC Ethernet card is already wired to the LAN then your routing table may have additional entries. These additional entries represent any routing devices that may be part of your network.

TCP/IP over the IBM Token Ring interface card, with no Ethernet card

1. Install the IBM Token Ring Network card using the instructions found in one of the New System Installation books. If cabling to the ring is available, you may also wire the Token Ring card to the MAU at this time.
2. Refer to the section covering LAN software package installation in the applicable CONVERSANT V6 Maintenance book to begin the software install. There you will ensure that all of the necessary UnixWare TCP/IP packages are loaded on the system. If they are not, follow the instructions to load TCP/IP from the UnixWare tape.
3. Complete the various software installation procedures for the entire Synchronous Host Interface feature. These procedures, located in the applicable CONVERSANT V6 Maintenance book , are the following:
   - — - “Installing the Token Ring Hardware Support Package”
   - — - “Installing the LINKix Packages”
   - — - “Installing the VIS Host Packages”
4. Following the software installation procedures, refer to the Novell UnixWare guide entitled “TCP/IP & NFS; TCP/IP Administration”. In chapter 3, pages 44-47, perform the following steps:

   Enter: **/etc/confnet.d/configure -i**
5. The configure command will prompt for the interface type. Select “ibmtok_0”.
6. The configure command will prompt for the host name. This is what is known as the network node name for the system, the name of your CONVERSANT system as you want it to appear in the /etc/hosts file. The typical response is to enter the system UNIX name or some alias name for the system.
7. The configure command will prompt for the ifconfig options. (The choices here are: yes, no, ClassC, BerkeleyC, info.) Select “ClassC”
8. From page 47 in the UnixWare TCP/IP Administration guide, change the route daemon to the active mode by editing the /etc/inet/rc.inet file and changing the line that looks like:

   ```
   /usr/sbin/in.routed -q
   ```

   to look like:

   ```
   /usr/sbin/in.routed
   ```
9. Enter: **shutdown -y -g0 -i6**
10. Edit the /etc/hosts file to enter the remaining network node names for your LAN.

11. Run the following command to display the network routing table:

    netstat -r

    The response should look something like this:

```
Routing tables
Destination  Gateway     Flags  Refs  Use  Interface
localhost    localhost   UH     0     0    lo0
xxx.yy.zzz   host_name   U      ?     ?    ibmtok0
```

12. If the Token Ring card is already wired to the MAU then your routing table may have additional entries. These additional entries represent any routing devices that may be part of your network.

TCP/IP over the IBM Token Ring interface card and SMC Ethernet card

1. Install the IBM Token Ring Network card using the instructions found in the appropriate System Installation book. If cabling to the ring is available, you may also connect the Token Ring card to the MAU at this time.
2. Complete the various software installation procedures for the entire Synchronous Host Interface feature. These procedures, located in the applicable CONVERSANT V6 Maintenance book, are the following:
   - — “Installing the Token Ring Hardware Support Package”
   - — “Installing the LINKix Packages”
   - — “Installing the VIS Host Packages”
3. Following the software installation procedures, refer to the Novell UnixWare guide entitled “TCP/IP & NFS; TCP/IP Administration”. In chapter 3, pages 44-47, perform the following steps:

   Enter: **/etc/confnet.d/configure -i**
4. The configure command will prompt for the interface type. Select “ibmtok_0”. If this choice is accepted, skip to step 8. If “ibmtok_0” is not one of the choices for configuration, continue with step 5.
5. Edit the following file and make an entry for “ibmtok_0” (here, the *vi* editor is used as an example):

   vi /etc/confnet.d/netdrivers
6. Add the following line to this file:

   ibmtok_0 inet
7. Re-run the **configure -i** command and select “ibmtok_0”.

8. The configure command will prompt for the host name. This is what's known as the network node name for the system, the name of your CONVERSANT system as you want it to appear in the /etc/hosts file. Remember, in this procedure we are connecting the CONVERSANT system to a second network. The ethernet LAN was configured previously, and now we are configuring a token ring LAN. Therefore, the token ring name for the system in the /etc/hosts file must be unique.
9. The configure command will prompt for the ifconfig options. (The choices here are: yes, no, ClassC, BerkeleyC, info.) Select "ClassC"
10. The configure command will also ask if you want to use this machine as a gateway. Respond accordingly.
11. From page 47 in the UnixWare TCP/IP Administration guide, check the /etc/inet/rc.inet file to see if the routing daemon is in the active mode. Ensure that the line looks like this:

    `/usr/sbin/in.routed`

    rather than this:

    `/usr/sbin/in.routed -q`
12. Now that TCP/IP has been configured on the IBM Token Ring card, you must either physically connect the card to an active MAU (and then skip to step 17), or if access to the ring is not available yet, you must edit the following file (here, the *vi* editor is used as an example).

    vi /etc/confnet.d/inet/interface
13. Move to the bottom of this file and comment out the line that begins with "ibmtok".

    If this change is not made to the /etc/confnet.d/inet/interface file, then the next time the CONVERSANT is rebooted, the system will try to access the token ring and fail. The following message will appear on boot up:

    "`TCP/IP start up not entirely successful. Error in /tmp/inet.start`"

    If this error message appears, the system will no longer be able to connect to the ethernet network, even though the SMC Ethernet card is still physically connected to the LAN and the card is fine.
14. If the IBM Token Ring card is not physically connected to an active ring stop here. When you have made the connection to the ring, continue with step 15 of the procedure.
15. Now that the IBM Token Ring card has been hooked up to an active MAU, you must edit the following file (here, the *vi* editor is used as an example).

    vi /etc/confnet.d/inet/interface
16. Move to the bottom of this file and comment in the line that begins with "ibmtok".
17. Enter: **shutdown -y -g0 -i6**

18. Edit the /etc/hosts file to enter the remaining network node names for your token ring LAN.
19. Run the following command to display the network routing table:

    netstat -r

    The response should look something like this:

```
Routing tables
Destination   Gateway      Flags   Refs   Use   Interface
localhost     localhost    UH      0      0     lo0
aaa.bb.ccc    host_name1   U       ?      ?     sme0
xxx.yy.zzz    host_name2   U       ?      ?     ibmtok0
```

20. If the ethernet LAN or the token ring network include any routing devices, then the routing table may have additional entries.

TCP/IP on the SMC Ethernet card but not on a Token Ring card installed for SNA-3270

1. Install the SMC Ethernet LAN card using the instructions found in the appropriate System Installation book. If cabling to the network is available, you may also wire the Ethernet card to the network hub at this time.
2. Refer to the section covering LAN software package installation in the applicable CONVERSANT V6 Maintenance book to begin the software install. There you will ensure that all of the necessary UnixWare TCP/IP packages are loaded on the system. If they are not, follow the instructions to load TCP/IP from the UnixWare tape.
3. Complete the installation procedure found in the applicable CONVERSANT V6 Maintenance book entitled “Installing the EtherCard Driver Package”.

   NOTE:
   The user responses provided in this section, for instance the IRQ, the I/O case address, the RAM address, the network connection (BNC, Twisted Pair), are simply recommendations. If your situation calls for a different IRQ or a BNC connection rather than twisted pair, then respond to the prompts accordingly.

4. Following the software installation procedure, refer to the Novell UnixWare guide entitled “TCP/IP & NFS; TCP/IP Administration”. In chapter 3, pages 44-47, perform the following steps:

   Enter: **/etc/confnet.d/configure -i**

5. The configure command will prompt for the interface type. Select “sme_0”.

6. The configure command will prompt for the host name. This is what is known as the network node name for the system, the name of your CONVERSANT system as you want it to appear in the /etc/hosts file. The typical response is to enter the system UNIX name or some alias name for the system.
7. The configure command will prompt for the ifconfig options. (The choices here are: yes, no, ClassC, BerkeleyC, info.) Select “ClassC”
8. From page 47 in the UnixWare TCP/IP Administration guide, change the route daemon to the active mode by editing the /etc/inet/rc.inet file and changing the line that looks like:

   `/usr/sbin/in.routed -q`

   to look like:

   `/usr/sbin/in.routed`
9. Edit the following file (here, the *vi* editor is used as an example):

   vi /etc/confnet.d/inet/interface

   Move to the bottom of this file and comment out the line that begins with “ibmtok”. This change is necessary to keep the SMC Ethernet connection working regardless if there is a connection to the IBM Token Ring card. If this change is not made to the /etc/confnet.d/inet/interface file, then the next time the CONVERSANT is rebooted, the system will try to access the token ring and fail. The following message will appear on boot up:

   “`TCP/IP start up not entirely successful. Error in /tmp/inet.start`”

   If this error message appears, the system will no longer be able to connect to the ethernet network, even though the SMC Ethernet card is still physically connected to the LAN and the card is fine.
10. Enter: **stop_vs**
11. Run smc_setup again to reaffirm the entries that you made during the EtherCard Driver installation.
12. Enter: **/etc/conf/bin/idbuild**
13. Enter: **shutdown -y -g0 -i6**
14. Edit the /etc/hosts file to enter the remaining network node names for your LAN.
15. Run the following command to display the network routing table:

    netstat -r

The response should look something like this:

```
Routing tables
Destination  Gateway     Flags  Refs  Use  Interface
localhost    localhost   UH     0     0    lo0
xxx.yy.zzz   host_name   U      ?     ?    sme0
```

16. If the SMC Ethernet card is already wired to the LAN then your routing table may have additional entries. These additional entries represent any routing devices that may be part of your network.

TCP/IP on the SMC Ethernet card with an IBM Token Ring card already running TCP/IP.

1. Install the SMC Ethernet LAN card using the instructions found in the appropriate System Installation book. If cabling to the network is available, you may also connect the Ethernet card to the network hub at this time.
2. Complete the procedure entitled “Installing the EtherCard Driver Package” found in the applicable CONVERSANT V6 Maintenance book.

   NOTE:
   The user responses provided in this section, for instance the IRQ, the I/O case address, the RAM address, the network connection (BNC, Twisted Pair), are simply recommendations. If your situation calls for a different IRQ or a BNC connection rather than twisted pair, then respond to the prompts accordingly.

3. Following the software installation procedure, refer to the Novell UnixWare guide entitled “TCP/IP & NFS; TCP/IP Administration”. In chapter 3, pages 44-47, perform the following steps:

   Enter: **/etc/confnet.d/configure -i**

4. The configure command will prompt for the interface type. Select “sme_0”. If this choice is accepted, skip to step 8. If “sme_0” is not one of the choices for configuration, continue with step 5.
5. Edit the following file and make an entry for “sme_0” (here, the *vi* editor is used as an example):

   vi /etc/confnet.d/netdrivers

6. Add the following line to this file:

   sme_0 inet

7. Re-run the **configure -i** command and select “sme_0”.
8. The configure command will prompt for the host name. This is what’s known as the network node name for the system, the name of your CONVERSANT system as you want it to appear in the /etc/hosts file.

Remember, in this procedure we are connecting the CONVERSANT system to a second network. The token ring network was configured previously, and now we are configuring an ethernet LAN. Therefore, the ethernet name for the system in the /etc/hosts file must be unique.

9. The configure command will prompt for the ifconfig options. (The choices here are: yes, no, ClassC, BerkeleyC, info.) Select "ClassC"
10. The configure command will also ask if you want to use this machine as a gateway. Respond accordingly.
11. From page 47 in the UnixWare TCP/IP Administration guide, check the /etc/inet/rc.inet file to see if the routing daemon is in the active mode. Ensure that the line looks like this:

    `/usr/sbin/in.routed`

    rather than this:

    `/usr/sbin/in.routed -q`
12. Now that 2 TCP/IP interfaces have been configured on the system, the IBM Token Ring card MUST remain physically connected to an active MAU. If the connection to the Token Ring is up and active, skip to step 15. If access to the ring is no longer available, continue with the next step in this procedure.
13. Edit the following file (here, the *vi* editor is used as an example):

    vi /etc/confnet.d/inet/interface
14. Move to the bottom of this file and comment out the line that begins with "ibmtok". If this change is not made to the /etc/confnet.d/inet/interface file, then the next time the CONVERSANT is rebooted, the system will try to access the token ring and fail. The following message will appear on boot up:

    "`TCP/IP start up not entirely successful. Error in /tmp/inet.start`"

    If this error message appears, the system will no longer be able to connect to the ethernet network, even though the SMC Ethernet card is still physically connected to the LAN and the card is fine.

    Now skip to step 17.
15. Edit the following file (here, the *vi* editor is used as an example):

    vi /etc/confnet.d/inet/interface
16. Move to the bottom of the file and make sure that the final entries in this file are in the following order:

    ```
    lo:0: ......
    sme:0:......
    ibmtok:0: ......
    ```

    If the entries are not in the proper order in this file, the network routing table may not build correctly.

17. Enter: **stop_vs**
18. Run smc_setup again to reaffirm the entries that you made during the EtherCard Driver installation.
19. Enter: **/etc/conf/bin/idbuild**
20. Enter: **shutdown -y -g0 -i6**
21. Edit the /etc/hosts file to enter the remaining network node names for your LAN.
22. Run the following command to display the network routing table:

    netstat -r

    The response should look something like this:

```
Routing tables
Destination   Gateway      Flags   Refs   Use   Interface
localhost     localhost    UH      0      0     lo0
aaa.bb.ccc    host_name1   U       ?      ?     sme0
xxx.yy.zzz    host_name2   U       ?      ?     ibmtok0
```

23. If the ethernet LAN or the token ring network include any routing devices, then the routing table may have additional entries.
24. If the IBM Token Ring card is disconnected (that is, if step 14 is used rather than step 16), then the network routing table will not contain any entries of the type “ibmtok0”

SQL*NET Communications

SQL*NET is the ORACLE communications component that allows the INTUITY CONVERSANT VIS to share information stored in different remote ORACLE databases. With SQL*NET, you can run an ORACLE tool or another application on the VIS and be able to find, manipulate, and store data in an ORACLE database located on another machine.

For additional information on ORACLE SQL*NET communications, refer to *ORACLE SQL*NET TCP/IP Documentation*, 585-350-913.

Asynchronous Communications

Asynchronous communications is a method of data transmission that allows characters to be sent at irregular intervals by preceding each character with a start bit and following it with a stop bit.

The VIS supports two standard asynchronous connections and one standard parallel printer connection on each of the Multi-Application Platforms (MAP) via an EIA-232 serial port. One of the standard asynchronous connections is reserved for the Remote Maintenance board. This circuit card provides a standard modular connection for access to the built-in modem. This arrangement allows access to the VIS through a remote terminal. This makes it possible to monitor system output and alarms, manipulate system resources, and perform software-related tasks without being physically near the VIS platform.

Data transmission is limited to 9600 bps (maximum) for asynchronous communication established with any device.

Refer to the section “Ports” and “Printers” in Chapter 8, “System Administration Features,” of the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591, for information on setting up the ports and printers.

Standard Asynchronous Connections

The standard asynchronous ports are located on the back of each MAP/100C, MAP/100, and MAP/40 unit. These connections and their locations are described for each MAP later in this section.

Note that the distance between transmission devices (for example, the VIS and a terminal) should not exceed 15 meters (50 feet) according to the EIA-232 standard recommendation. Devices can be separated by longer distances, however, depending on how much electrical interference exists in the area. Use an asynchronous data unit (ADU)for distances from 15 to 1525 meters (50 to 5000 feet). Refer to the appropriate ADU documentation for maximum limits.

MAP/100C Asynchronous Communication Ports

You can connect the MAP/100C platform to a terminal, modem, or host computer through an asynchronous link connected to one or more ports on the platform. The system is connected to a printer through a single parallel port. The standard connections include:

- Two 25-pin D-subminiature male port, COM2, located on the ront and rear lower center of the MAP/100C. Both connectors provide access to the COM2 port on the CPU, for convenient access to central office rack-mounted systems.

  NOTE:
  The Remote Maintenance board uses the COM2 serial port.

- A 9-pin D-subminiature male port, COM1, is located on the faceplate of the CPU circuit card, which is accessed by opening the card cage access door.

NOTE:
The COM1/COM2 orientation is different between MAP/100C and the commercially available MAP/100 and MAP/40 machines. On the MAP/100C, the CPU-mounted connector is labeled "COM2," while the externally wired connector(s) are labeled “COM1.” This is reversed for the MAP/100 and MAP/40 machines.

- An Asynchronous Communications Interface feature package is available on the MAP/100C to provide eight additional RJ-45 type modular connector asynchronous ports. In earlier versions of the CONVERSANT system, this package included an optional circuit card, software, and T-adapter. Version 6.0 includes an 8-port asynchronous circuit card and a software driver.
- A parallel port connection is located on the faceplate of the CPU card. This port is a 25-pin male connector and is used as a printer interface.

MAP/100 Asynchronous Communication Ports

You can connect the MAP/100 platform to a terminal, modem, or host computer through an asynchronous link connected to one or more ports on the platform, including:

- A 9-pin D-subminiature male port, COM1, located on the faceplate of the CPU circuit card

  NOTE:
  The Remote Maintenance Circuit Card uses the COM1 serial port.

- A 9-pin D-subminiature male port, COM2, located at the rear, upper left corner of the MAP/100 chassis
- An Asynchronous Communications Interface feature package is available on the MAP/100 to provide eight additional RJ-45 type modular connector asynchronous ports. In earlier versions of the CONVERSANT system, this package included an optional circuit card, software, and T-adapter. Version 6.0 includes an
  8-port asynchronous circuit card and a software driver.
- A parallel port connection is located on the faceplate of the CPU card. This is a 25-pin male connector and is used as a printer interface.

MAP/40 Asynchronous Communication Ports

The MAP/40 platform can be connected to a terminal, modem, or host computer through an asynchronous link connected to one or more ports on the platform, including:

- A 9-pin D-subminiature male port, COM1, located on the faceplate of the CPU circuit card

NOTE:
The Remote Maintenance Circuit Card uses the COM1 serial port.

- A 9-pin D-subminiature male port, COM2, located at the rear, middle right side of the MAP/40 chassis
- An Asynchronous Communications Interface feature package is available on the MAP/40 to provide eight additional RJ-45 type modular connector asynchronous ports. In earlier versions of the CONVERSANT system, this package included an optional circuit card, software, and T-adapter. Version 6.0 includes an 8-port asynchronous circuit card and a software driver.
- A parallel port connection is located on the faceplate of the CPU card. This is a 25-pin male connector and is used as a printer interface.

8-Port Asynchronous Circuit Card Connections

Each of the MAP systems support connections to one or more asynchronous host computers or additional modems via an 8-port asynchronous interface. Depending on the version of the platform, these eight additional serial ports are provided by either an IPC-900, Gemini 1000, or Equinox 8-port asynchronous circuit card.

These serial connection ports are configured as data terminal equipment (DTE). DTE ports require a crossover or “null modem” cable to connect to serial devices such as a terminal, computer, or printer. The term “crossover” refers mainly to the transmit and receive lines. To communicate with any of the devices mentioned above, the transmit line on the serial port must ultimately be connected to the receive line of the terminal device. Conversely, the receive line on the serial port must be connected to the transmit line of the terminal device.

Connecting to a modem does not require a crossover cable. A modem is normally considered data communications equipment (DCE). DCE ports require a modem or straight-through cable. The crossover of transmit and receive are handled within the modem.

The following adapters are available to allow DCE equipment to communicate with DTE and vice versa:

- Null modem adapter or cable. This adapter “flips” the transmit and receive lines while still maintaining the functions of the other lines, that is, data terminal ready (DTR) and ground. This device is normally used to connect one DTE device to a another DTE device.
- Terminal/printer adapter. This adapter provides a crossover function much the same as a null modem adapter.
- ACU modem adapter. This is an adapter or cable that provides a straight-through connection.

- Gender changers. Gender changers convert a male connector to female and vice versa. There are two types of gender changers, male/male and female/female. The functionality of the incoming lines is maintained on the outgoing side.
- Modular extenders. Extenders allow you to connect two modular cables to each other without losing functionality. An extender consists of two female RJ-45 type ports linked to each other. The number of conductors in the extender must match the number of conductors in the cables used. There are three types of modular cables used with asynchronous communications within the VIS:
  - A 6-conductor telephone hook-up cable (three pair) for analog Tip/Ring (T/R) connections.
  - An 8-conductor cable is used for serial port peripheral connections (the standard serial ports provided on each VIS platform)
  - A 10-conductor cable is used to connect devices with the modular ports provided on the 8-port asynchronous circuit card.

It is possible to connect 8-conductor to 10-conductor cables. The adapters used with the 8-conductor cable must be 8-pin adapters. Ten-pin adapters can be used with 10-conductor modular cables only. Eight-pin adapters can be connected to 10-pin adapters. However, check the wiring diagrams of both adapters to make sure that there is not loss of functionality when connecting 8- to 10-pin adapters.

In most cases, if transmit goes to receive (and vice versa) in connecting DTE devices, any combination of equipment can be used. For modems, it is most likely that a straight-through connection is required since they are DCE devices. However, you should confirm the pin positions of other functions (that is, DSR, DTR, carrier, etc.) on all connected devices to ensure proper functionality.

8-port Asynchronous Connections to Terminals

Figures 7-8 and Figure 7-9 show examples of external connectivity and cabling for a 8-port asynchronous connection to a terminal. Note that these are only examples and not an exhaustive list of possible connections.

NOTE:
The type of conductor cable (6- or 10-conductor) depends on 8-port asynchronous unit (Equinox, IPC900, or Gemini 1000) from which you are making the connection. Refer to the Appendix B, “Cable Connectivity,” of the hardware installation book for your platform for a list of the parts required.

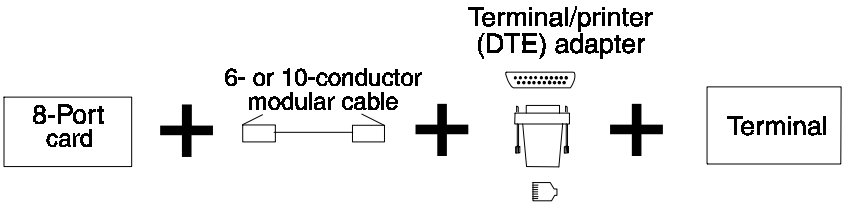


Figure 7-8. 8-port Asynchronous Terminal Connection Using 6- or 10-Conductor Modular Cable

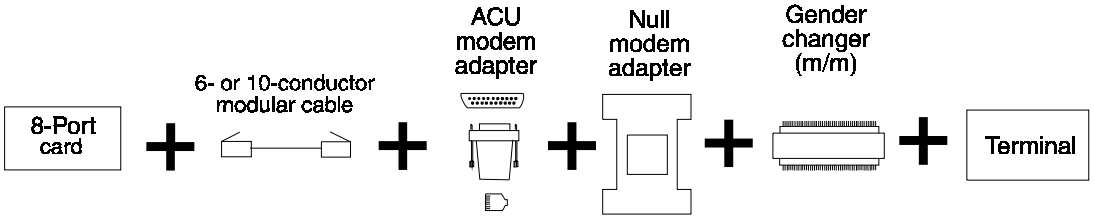


Figure 7-9. 8-port Asynchronous Terminal Connection Using 6- or 10-Conductor Modular Cable and a Null Modem

8-port Asynchronous Connections to Computers

Figures 7-10 and 7-11 show examples of external connectivity and cabling for a multi-port asynchronous connection to a computer. Note that these are only examples and not an exhaustive list of possible connections.

NOTE:
The type of conductor cable (6- or 10-conductor) depends on 8-port asynchronous unit (Equinox, IPC900, or Gemini 1000) from which you are making the connection. Refer to the Appendix B, “Cable Connectivity,” of the hardware installation book for your platform for a list of the parts required.

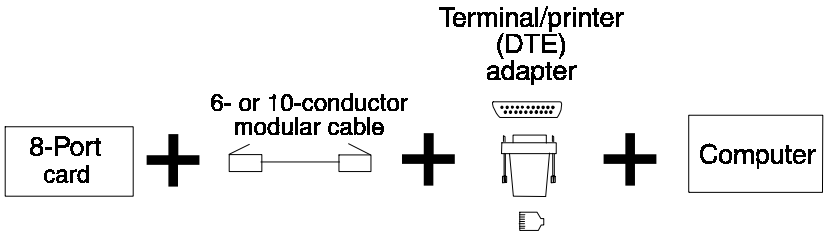


Figure 7-10. 8-port Asynchronous Computer Connection Using 6- or 10-Conductor Modular Cable

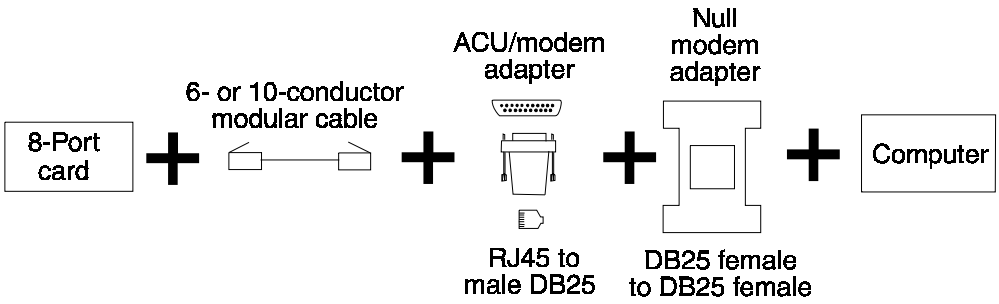


Figure 7-11. 8-port Asynchronous Computer Connection Using 6- or 10-Conductor Modular Cable and a Null Modem

8-port Asynchronous Connections to an External Modem

Figure 7-12 and Figure 7-13 show examples of external connectivity and cabling for a multi-port asynchronous connection to an external modem. Note that these are only examples and not an exhaustive list of possible connections.

NOTE:
The type of conductor cable (6- or 10-conductor) depends on 8-port asynchronous unit (Equinox, IPC900, or Gemini 1000) from which you are making the connection. Refer to the Appendix B, “Cable Connectivity,” of the hardware installation book for your platform for a list of the parts required.

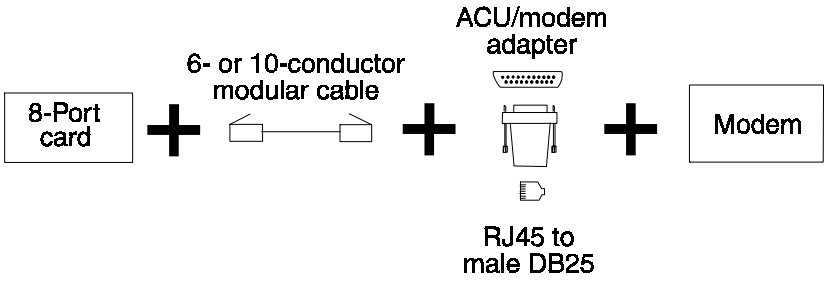


Figure 7-12. 8-port Asynchronous Modem Connection Using 6- or 10-Conductor Cable

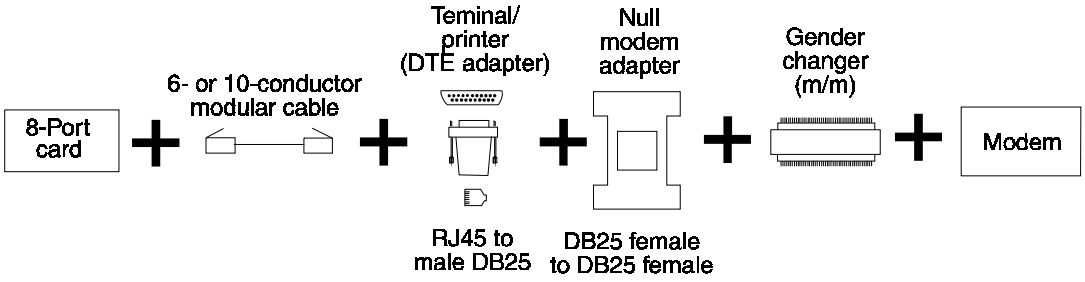


Figure 7-13. 8-port Asynchronous Modem Connection using 6- or 10-Conductor Cable and a Null Modem

8-port Asynchronous Connection to an ADU

Figure 7-14 shows an example of external connectivity and cabling for an 8-port asynchronous connection to an ADU.

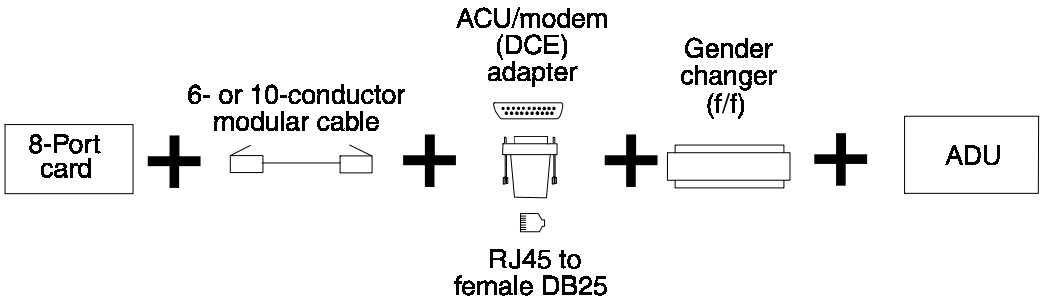


Figure 7-14. 8-port Asynchronous ADU Connection

8-port Asynchronous Connection to a Printer

There are two ways to connect the 8-port asynchronous unit to a printer. If you are connecting to the serial port on the printer, connect the 8-port card the same way that you would connect to a terminal as shown in Figure 7-8. You can use a DTR printer adapter in place of the DTE terminal/printer adapter if the software uses the hardware flow control for the specified port.

Data Network Connectivity Alarms

8

What's in This Chapter

The following data network alarming packages are available for use in conjunction with the INTUITY CONVERSANT Voice Information System V6.0 software:

- NetView
- CompuLert/SCCS
- External Alarms

This chapter provides information on each of these packages, including configuration and administration procedures.

NetView Alarming

The NetView Alarming software package interacts with the INTUITY CONVERSANT System V6.0 software to allow you to monitor system messages as part of your current NetView environment. The CONVERSANT System logs alarms and events that occur during voice system operations. The system's maintenance transmitter (mtcxmtr) process scans this log for error conditions and transmits critical, major and minor errors to the host as Operator-Generated Alerts (OGAs) over the 3270 host link.

Refer to the *NetView User's Guide* that is a part of the *CLEO LINKix Documentation Set*, 585-310-912, for information on accessing the NetView program, using the NetView screen display and commands, and using the Network Management Application Programing Interface (NM-API).

Configuring NetView

The NetView Alarming package is now bundled with the host interface offer (refer to Chapter 7, “Data Network Communications”). If you do not want NetView alarms sent to the host, remove the INTUITY CONVERSANT System V6.0 3270 NetView Alarm Interface package. Refer to , “Installing Optional Features,” in the appropriate *Intuity* ™ CONVERSANT® System Version 6.0 maintenance book.

By default, the host interface software is configured at installation to send all NetView alerts over the first host connection that is defined. To change the connection over which the alarms are sent

1. Enter **hconfig -a *conn_name***, where *conn_name* is the name of the connection you want to use.
2. Stop and start the voice system. Refer to “Common Maintenance Procedures,” in the appropriate *Intuity* ™ CONVERSANT® System Version 6.0 maintenance book.

When migrating to an environment with different NetView requirements, you must reconfigure NetView by editing the NetView configuration file. This configuration file contains flag settings that are used for configuring, monitoring, and testing the maintenance transmitter. The values set at installation should be satisfactory for normal operation in most environments. However, some environments may require a modification of some of these configuration flags.

The maintenance transmitter reads the configuration file when it starts up or when it receives the signal SIGUSR2. It is therefore possible to change the behavior of the maintenance transmitter by either sending it a SIGUSR2 signal or using it to restart and automatically read the configuration file.

NOTE:
For either of the following approaches, you must use the Process ID number (PID). Enter **ps -ef | grep mtcxmtr** to determine the PID. The PID is the leftmost number displayed on the output.

Changes in the configuration file take effect when you do one of the following:

- Enter **kill -17 *<PID>*** where *<PID>* is the process id number to send a SIGUSR2 signal to the maintenance transmitter. This approach is nondisruptive to the system (for example, queued alarms are saved).
- Enter **kill -9 *<PID>*** where *<PID>* is the process id number to restart the maintenance transmitter. This approach causes any accumulated error messages to be lost, while maintaining the host connection. The advantage of this approach is that it will start a new log, making it easier to locate the results of subsequent tests.

  NOTE:
  When you restart the maintenance transmitter, the previous log that records the transmitter’s actions is moved to **/tmp/al_log.old**.

Configuration information is stored in an ASCII file called **/vs/data/alarm_flags**. To view or edit the contents of this file, log in as root and enter **vi /vs/data/alarm_flags**

Figure 8-1 shows the default values in the **/vs/data/alarm_flags** file. A description of each flag in the file follows the example.

```
Setbuf=1
Verbose=3
Tst=0
fail_mod=10
Wrap_ct=20,Rate=400
```

- Setbuf

  The Setbuf flag indicates whether messages sent to the log file will be buffered. A nonzero integer setbuf value indicates that messages will not be buffered.

  NOTE:
  Unbuffered writes to the log file are less efficient than buffered writes. Use unbuffered writes only when you want to guarantee the integrity of the log file through a system crash.

- Verbose

  The Verbose flag indicates the level of detail of the maintenance transmitter activity in the log. The following details the possible Verbose settings.

-1	No logging
3	Basic actions (default)
7	Detail of structure sent to the host
10–25	Increasing amount of detail in the log

  NOTE:
  The default value of 3 is strongly recommended for normal operation. The log is normally written without buffering from the UNIX operating system.

- Tst

  The Tst flag indicates the mode of operation. A Tst setting of 0 indicates remote operation. A Tst setting of 1 indicates local testing. A Tst setting of 2 indicates local testing but generates the maximum number of OGAs storable when any are read.

- fail_mod

  The fail_mod flag indicates the fraction of transmissions that will fail in test mode. The fail_mod flag causes the maintenance transmitter to simulate one transmission to fail. For example, if fail_mod is 4, an average of 25% of the transmissions will fail, with the transmissions of that file being randomly selected. The fail_mod flag if used only in test mode (that is, with Tst set to 1 or 2).

- Wrap_ct

  The Wrap_ct flag is used along with the Rate flag (described below) to define the maximum number of OGAs that can be transmitted. In the production environment (normal operation), set Wrap_ct to 10. In the test environment, set Wrap_ct to 20. To ignore Wrap_ct constraints established by NetView, set Wrap_ct to 0.

- Rate

  The Rate flag is used along with the Wrap_ct flag to define the maximum number of OGAs that can be transmitted. In the production environment (normal operation), set Rate to 1200. In the test environment, set Rate to 400. To ignore Rate constraints established by NetView, set Rate to 0.

  The Wrap_ct and Rate flags are used in the following manner: when an OGA arrives, NetView verifies that the maximum number of OGAs (as defined by the Wrap_ct flag) arrived in the allowable number of seconds (as defined by the Rate flag). For the test environment, Wrap_ct is 20 and Rate is 400. Consequently, when an OGA arrives in the test environment, the 20th previous OGA should have arrived not less than 400 seconds ago. For the production environment, Wrap_ct is 10 and Rate is 1200. Consequently, when an OGA arrives in this environment, the 10th previous OGA should have arrived not less than 1,200 seconds ago.

  NOTE:
  The host interface card (the PC/XL and/or the FIFO-SIB) accepts OGAs at a maximum average rate of approximately one per second. Within these limits, the maintenance transmitter sends OGAs as soon as possible in the first-in, first-out order.

Testing the Maintenance Transmitter

To test the maintenance transmitter, instruct the CONVERSANT System software to send a known set of error messages to it and observe the resulting OGAs.

1. Use the command **logit** to generate error messages. For example, you can construct a script that contains the following lines to drive NetView testing:

```
logit -p minor -d 0xffff "This is test message 1"
logit -p minor -d 0xffff "This is test message 2"
logit -p minor -d 0xffff "This is test message 3"
logit -p minor -d 0xffff "This is test message 4"
logit -p minor -d 0xffff "This is test message 5"
```

   Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for additional information on using the **logit** command.

2. View the transmitter's output either by using NetView (if there is a live host connection) or by examining the log **/tmp/al_log**. Input to the maintenance transmitter can be determined by examining the log.

   The maintenance transmitter should pass the following tests:

   - Every logged message or priority — Critical, Major, and Minor — should generate an OGA.
   - Only error messages of priority Critical, Major, and Minor should generate OGAs.
   - The maintenance transmitter should follow the NetView constraints as expressed in the Wrap_ct/Rate line of the configuration file.
   - If the connection to the host is lost, the transmitter should check the link at 5-minute intervals and resume sending messages 5 minutes after the connection is reestablished. If the maintenance transmitter receives up to 100 OGAs, these messages are stored and sent at a later time. If the transmitter receives more than 100 OGAs, the messages that were received first are overwritten and lost. Consequently, when the link is restored, not all of the messages received while the link was down are sent.
   - OGAs should be in a format defined in "Configuring NetView Alarms" earlier in this chapter.

   NOTE:
   You can test everything but the NetView constraints in local mode. If you test in local mode, however, the failure rate due to the host interface card blocking may not be realistic.

   Figure 8-1 shows the maintenance transmitter sending five OGAs to the host computer. The text of the set of alarms to send is enclosed by the SEND_2_Host and N alarms in the alarmbuf line.

```
                SEND_2_HOST:  Feb 22 04:48:21

0 <*  LOGIT GEN002 -- -- --- root: This is test message 1 04:55>:
 Feb 22 04:58:21

1 <*  LOGIT GEN002 -- -- --- root: This is test message 2 04:55>:
 Feb 22 04:58:21

2 <*  LOGIT GEN002 -- -- --- root: This is test message 3 04:55>:
 Feb 22 04:58:21

3 <*  LOGIT GEN002 -- -- --- root: This is test message 4 04:55>:
 Feb 22 04:58:21

4 <*  LOGIT GEN002 -- -- --- root: This is test message 5 04:55>:
 Feb 22 04:58:21


                   5 alarms in alarmbuf:  Feb 22 04:58:21
```

Figure 8-1. Maintenance Transmitter Log Example

External Alarms

The External Alarms Interface package provides a means for administering external alarms in a central office environment. This package can be used only on the MAP/100C. This section describes how to provision the External Alarms feature for monitoring the CONVERSANT System software on a MAP/100C.

External Alarms Relay Contacts

To provision this feature correctly, you must understand the relay contacts used by the External Alarms Interface card. The External Alarms Interface card includes relays controlled by a sanity timer, by power failure, and by the CONVERSANT System software. The relays can alert receptors external to the CONVERSANT System of problems within the system.

Sanity Timer Relay Contacts

The Sanity Timer controls relay number 8 or relays 7 and 8 depending on how the card is configured (refer to the hardware installation book for your platform for additional information). The Sanity Timer is used to indicate that the software on the CONVERSANT System is running. The Sanity Timer must be reset periodically by a process within the system. As long as the sanity timer is reset by this process, the Sanity Timer will not time out, and the relay(s) associated

with the timer will not close. The sanity timer is updated by the alerter process on the CONVERSANT System. The alerter process runs at run-level 2 so the Sanity Timer will not time out even if the voice system is stopped. The most likely cause of Sanity Timer time-out is a system crash or a system lockup.

Power Fail Relay Contact

The Power Fail Relay Contact, relay 1 on the External Alarm Interface card, remains closed as long as there is power to the External Alarm Interface card. Power comes to the card from the CONVERSANT System backplane. The Power Fail Relay Contact opens if power is cut off from the CONVERSANT System, or if the External Alarm Interface card is not seated properly in the backplane. There is no software control available for the Power Fail Relay Contact.

Software-Controllable Relay Contacts

The remaining relay contacts are software controllable. That is, the CONVERSANT System resident software can send commands to the External Alarm Interface card to open and close relay contacts. Software controllable relay contacts include relays 2 through 6 and possibly 7 depending on how the card has been configured (refer to the hardware installation book for your platform for additional information).

External Alarms Interface Software Features

The primary function of the software supplied with the External Alarms Interface Package is to close relay contacts when the CONVERSANT System generates certain alarm-level messages. The software supports mapping messages to one or more relay contacts, or none at all. The software also provides an administrative command set. This command set supports the enabling and disabling of message-produced relay contact closures and state changes to the relay contacts themselves.

Software Interface to the External Alarms Interface Card

The software implementing the External Alarms Interface consists of a process that monitors system messages (the alerter) and a command set. The alerter is also responsible for updating the sanity timer at a regular interval. (The default is every 20 seconds.) The alerter uses a notion of Alarm Contact Sets. An Alarm Contact Set is a set of software controllable relay contacts. The file **/vs/data/alarms/masks** specifies the External Alarms card relays associated with a given Alarm Contact Set.

System messages are then assigned to Alarm Contact Sets through inclusion in one or more of the alarm files in **/vs/data/alarms**. For example, all messages assigned to Alarm Contact Set 1 are specified in **/vs/data/alarms/alarm1**.

When the system generates a message, the alerter reads it. If its ID is in one of the alarm files, the Alarm Contact Set associated with that file is closed. Note that a message ID can reside in more than one alarm file.

Note that Alarm Contact Sets and External Alarms Card relay contacts are not necessarily the same thing. Alarm Contact Sets provide a level of indirection between the software and the hardware. This allows more than one External Alarms Card relays to be assigned to a single Alarm Contact Set, and it allows more meaningful numeric identifiers to be associated with the relays. For instance, with the defaults settings, Critical, Major and Minor alarms are assigned to alarm contact sets 1, 2, and 3, respectively. However, Alarm Contact Sets 1, 2, and 3 map to Alarm Card relays 6, 5, and 4, which is nonintuitive. See “Mapping Alarm Contact Sets to Alarm Card Relays” later in this chapter.

External Alarms Connectivity

Figure 8-3 shows a possible External Alarms central office configuration. In this example, a machine alarm light is illuminated for the Sanity Time (Relay 8) as well as for Critical, Major, or Minor alarm occurrences (Relay 2). In addition, an aisle alarm grid is illuminated for the Sanity Timer (Relay 7), Critical (Relay 6), Major (Relay 5), and Minor (Relay 4) alarm occurrences. Relay 3 is unused in this configuration. The Grid Power, Sanity Time, or critical alarm lights the grid Critical light. The major alarm lights the grid Major light and the minor alarm lights the grid Minor light.

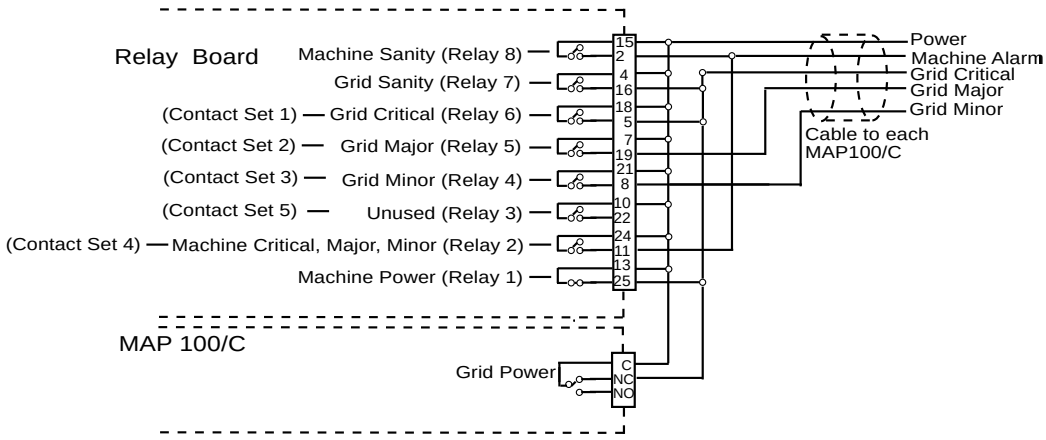


Figure 8-2. Alarm Relay Card Configured for a Central Office Application

External Alarms Administration

External Alarm Operational Commands

The External Alarms Interface package is delivered with a software command set for administration of the Alarm Contact Sets. The command set is implemented in the command **alarm**, which is executable from the UNIX system prompt. The External Alarms Interface provides the capability to enable, disable, display, reinitialize (reinit), retire, or test external alarms using the commands included with the External Alarms Interface package. Refer to the *Intuity* ™ CONVERSANT® System Version 6.0 *Administration* 585-310-591, for additional information on each of these commands.

The External Alarms Interface enable and disable features allow you to enable or disable a specified Alarm Contact Set. By default, all Alarm Contact Sets are set to enable; that is, the Alarm Contact Set is operational. Note that if the Alarm Contact Set is enabled, the contacts close upon receiving an assigned message or performing the alarm test command.

The **alarm display** command allows you to display the state of the external Alarm Contact Sets. The external Alarm Contact Sets are either OFF (the contacts are open indicating that no assigned message has occurred) or ON (the contacts are closed due to the occurrence of an assigned message).

Mapping Alarm Contact Sets to Alarm Card Relays

The software delivered with the External Alarms package provides a mapping from Alarm Contact Sets to Alarm Card Relays. When the software is installed, Alarm Contacts are assigned as follows:

alarm contact set 1:Alarm Card Relay 6
alarm contact set 2:Alarm Card Relay 5
alarm contact set 3:Alarm Card Relay 4
alarm contact set 4:Alarm Card Relay 2

It is possible that your application may require a different mapping. You can change the mapping of alarm contact sets to alarm card relays by editing the **/vs/data/alarms/masks** file.

The basic format of this file is:

<alarm contact set> <relay> [<relay>] ...

where *<alarm contact set>* is of the form alarm *X* and *X* is a single-digit number (for example: 1, 2, 3, ...) and *<relay> [<relay>]* ... is one or more software controllable alarm card relay numbers (2, 3, 4, 5, and/or 6)

Note that alarm contact sets must be disjointed; that is, two Alarm Contact Sets may not contain the same alarm relay number.

Note that there must be a **/vs/data/alarms/alarm*X*** file for each **alarm*X*** defined in **/vs/data/alarms/masks**. The following shows a **/vs/data/alarms/masks** default file:

```
alarm1    6
alarm2    5
alarm3    4
alarm4    2
```

Another possible **/vs/data/alarms/masks** file could be as follows:

```
alarm1    6 5
alarm2    4 3
alarm3    2
```

In this scheme, if a message is generated that has an ID is in **/vs/data/alarms/alarm1**, Alarm Contact Set 1 is set; that is, Alarm Card Relays 6 and 5 are closed.

Setting the Sanity Timer Update Time

The External Alarms card is equipped with a sanity timer. The sanity timer is used to inform you that the voice system may have stopped operating. This timer must be reset before it times out or relays 7 and 8 will close. The sanity timer is reset every 20 seconds by default. The voice system periodically accesses the External Alarms card to reset the sanity timer. To change the reset period of the external alarms software, place a time-out value in the file **/vs/data/alarms/timer**. This value should be a positive integer that represents a number of seconds. To change the update time to every 30 seconds, change the 20 to 30 in the **/vs/data/alarms/timer** file. In the absence of a **/vs/data/alarms/timer** file, the CONVERSANT System uses a 20-second time-out value.

It is also possible to change the time-out value on the External Alarms card. See the hardware installation book for your platform for information.

NOTE:
The software time-out value should be less than the hardware time-out value.

Voltage and Current Capacities for External Alarms Interface

Table 8-2 provides the current capacities for the External Alarms Interface hardware. If an inductive or capacitive load is being switched, the

- Peak turn-off or turn-on surge current must not exceed the DC current limit.
- Maximum AC or DC root mean square (RMS) load current must be less than the AC or DC current limit.

Table 8-1. Current Capacities for External Alarms

Voltage	Current (A)
250 VAC	5
30 VDC	5
125 VDC	1

Transmission Level Adjustment

A

Transmission Level Plan

A Transmission Level Plan (TLP) for a piece of telecommunications equipment is a set of specifications dictating the incoming/outgoing speech volume levels that pass through the equipment and the hardware and software tools for implementing those specifications. The specifications take into account the level plans of the various telephone network interfaces to which the equipment will connect. The goal of the plan is to ensure that all speech heard by a caller be at a level which is appropriate for listening without causing oscillations or distortions in the network.

CONVERSANT Network-Interface Hardware

The INTUITY CONVERSANT System connects to two types of telephone network facilities: analog (T/R) and digital (T1/E1).

The INTUITY CONVERSANT System's default TLP is partially based on the following facts concerning VIS network interface hardware:

- The system's T1/E1 interface circuit cards have a gain of 0 dB built into the hardware interface.
- The system's T/R interface circuit cards have a nominal gain of 0 dB built into the hardware interface (when a perfect impedance match exists between the interface and the line to which it is connected).

Typical Network TLP Characteristics

The T/R and T1/E1 network facilities have typical TLP characteristics associated with them. The system default TLP is partially based on the following typical network TLP characteristics:

- The system default TLP assumes a nominal 0-dB gain in each digital trunk connected to any T1/E1 card in the system.
- The system default TLP assumes a nominal -3-dB gain in each analog loop connected to any T/R card in the system.

Incoming and Outgoing Speech Volume Nonbridging Modes

When a voice signal enters a CONVERSANT machine in a nonbridged connection, it is usually going to be coded and stored in the speech filesystem of the machine. Before it is coded, its incoming volume can be adjusted by the IVOL parameter.

By default, all coding modes will be subjected to an automatic gain control (AGC) after the IVOL is applied. The AGC is used to maintain a proper recording level. AGC attenuates signals that would otherwise be too loud and amplifies signals that would otherwise be too quiet. For this reason, small adjustments of IVOL have little impact when AGC is active. It may, however, be necessary to increase IVOL if the input is so low the AGC takes it to be silence. (Such input the AGC treats as background noise and, for the listener's comfort, does not pass it. Consequently, input that is too low may be cut off and short phrases may be completely missing.)

When a voice signal stored in the speech file system is played back from a CONVERSANT machine to a caller, its outgoing volume can be adjusted by the OVOL parameter.

The INTUITY CONVERSANT Digital Interfaces screen allows the user to adjust both the incoming and outgoing speech volume for analog (T/R) and digital (T1/E1) network interfaces. The analog IVOL and OVOL parameters apply to all analog circuit cards in the system. The digital IVOL and OVOL parameters apply to T1/E1 circuit cards on a per card basis.

IVOL and OVOL should be thought of as volume multipliers (that is, +/- gain) of the incoming/outgoing signal. A value of 1000 for IVOL or OVOL is equivalent to multiplying the incoming or outgoing signal volume by 1,that is, *unity gain*. Each multiplication of the current IVOL or OVOL setting by a factor of 0.707 results in a -3 dB signal volume gain from the current volume (volume 3 dB lower); each multiplication of the current IVOL or OVOL setting by a factor of 1.414 results in a +3 dB signal volume gain from the current volume (volume 3 dB higher).

NOTE:
IVOL and OVOL affect only signals being coded or played back by the CONVERSANT System. They do not affect end-to-end conversations in call bridge mode, DTMF tone detection, CPT tone detection or speech recognition.

Table A-1 shows the IVOL and OVOL settings required to implement the default TLP along with the actual gain in decibels (shown in parenthesis) that each setting represents.

Table A-1. Default System IVOL and OVOL Settings

Network Facilities	IVOL	OVOL	Text-to-Speech (TTS) OVOL[1]
Analog	4000(+12)	1000(0)	4000
Digital	1414(+3)	707(-3)	1000[2]

1.The TTS OVOL is an option only when the TTS package is installed.
2.The TTS OVOL default value may be too low in some cases. You may want to use a higher value. However, if a value is too high, it may cause distortion of the outgoing text.

Voice Coding and Play

As described above, most switches build in some loss in a typical station-set-to-station-set connection. With the system in a nonbridging mode, station-set-to-station-set actually involves a signal being affected by IVOL while it is coded and stored on the disk, then affected by OVOL when it is played back. To be in accordance with the TLP, the level the caller hears during playback should be somewhat lower than the level that was spoken when the signal was coded. Considerations used to determine proper input and output volume are given under "Reasons for Deviating from the Default IVOL and OVOL Settings" on page A-5.

Voice Coding

Figure A-1 offers an example of how the IVOL parameters control the level at which a voice signal is coded and stored in the VIS speech filesystem. The levels in Figure A-1 illustrate the interaction between a switch and the INTUITY CONVERSANT System. (The actual default IVOL is +12dB rather than the +6 shown in the example. The +12dB level reduces the chance of low input volume levels being recored as silence. Automatic gain control (AGC) makes it unlikely that the higher input volume will cause clipping or other distortion. See "Reasons for Deviating from the Default IVOL and OVOL Settings" on page A-5 for details.)

The top part of Figure A-1 shows a T1/E1 interface connected to the VIS; the bottom part shows a T/R interface connected to the VIS. As you follow the signal from left to right, if the initial spoken level is 0 and all typical network TLP characteristics listed above are true, the coded level that is stored in the speech filesystem will always be zero (0), regardless of which type of network interface is connected to the VIS.

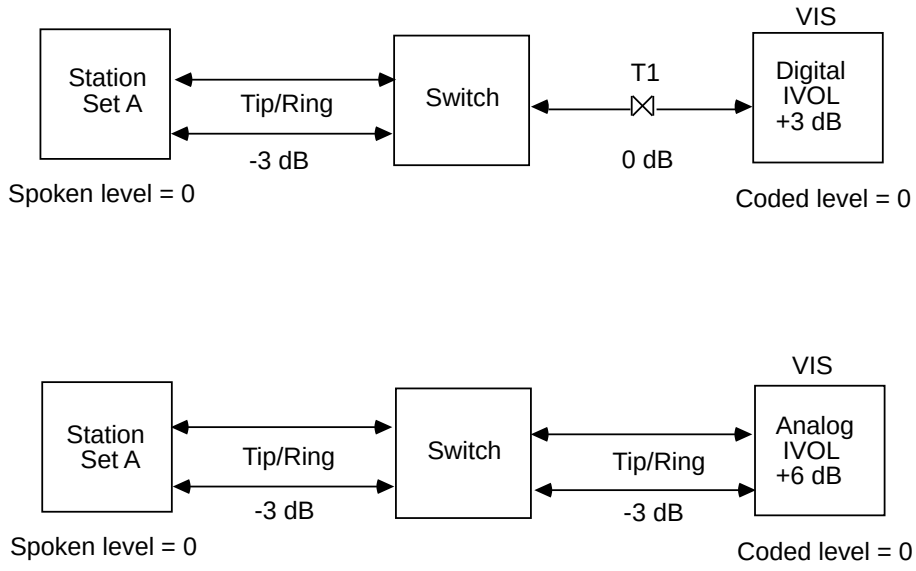


Figure A-1. Effect of IVOL Parameters Voice Coding

Voice Play

Figure A-2 shows how the default OVOL parameters control the level at which a previously coded voice signal stored in the VIS speech filesystem is played back.

The top part of Figure A-2 shows a T1 interface connected to the VIS; the bottom part shows a T/R interface connected to the VIS. As you follow the signal from right to left, if the signal was coded in the manner depicted in Figure A-1, the initial playback level is 0. If all typical network TLP characteristics listed above are true, the level heard at the station set is always -6, regardless of which type of network interface is connected to the VIS. Since the initial spoken level shown in Figure A-1 was 0, the heard level of -6 is in accordance with the VIS TLP.

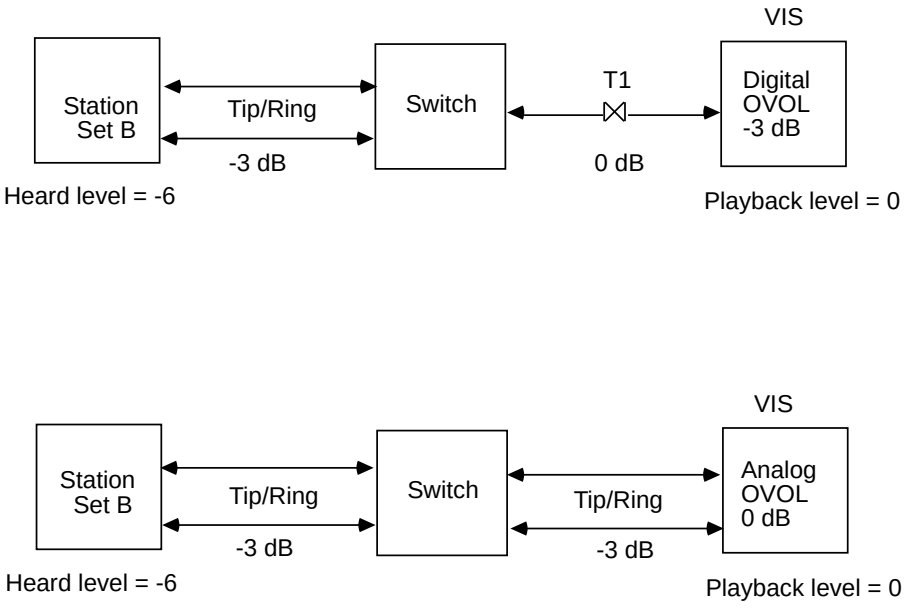


Figure A-2. Effect OVOL Parameters Voice Play

Reasons for Deviating from the Default IVOL and OVOL Settings

For most applications, the default TLP provides callers with appropriate speech volume levels for prompts that were coded as shown in Figure A-1.

In many cases, however, speech prompts are coded in a studio at higher volumes than they would have been coded from a VIS network interface. In these situations, it may be desirable to decrease the applicable OVOL parameter (analog or digital, depending on whether playback is from T/R or T1) to decrease the volume the caller actually hears. Note that if the system is used to code speech that will be played back with the prerecorded speech, you should increase IVOL by the same amount that you decrease OVOL to ensure that speech is coded at the same level.

Also, some network lines and/or trunks do not abide by the typical network characteristics listed above. For example, some T1 trunks actually have insertion loss in the network. This loss can be compensated for by increasing the corresponding IVOL and OVOL parameters by an amount equal to the additional insertion loss. For example, if the digital trunks connected to a VIS had insertion loss of -3 dB instead of 0 associated with them as the default CONVERSANT System TLP assumes, the default digital IVOL and OVOL parameters could be changed to 2000 and 1000, respectively. This would have the effect of adding a

gain of +3 dB to the incoming signal before coding, and adding a gain of +3 dB to the outgoing signal before playback (refer to Table A-1 and the accompanying explanation). Making these changes results in meeting the TLP goal of -6 dB gain from end to end.

If the IVOL is set too low, phrases may be cut short or may be missing. In such cases the input may be so low the automatic gain control (AGC) takes it to be silence. (Such input the AGC treats as background noise and, for the listener's comfort, does not pass it. Consequently, input that is too low may be cut short and some phrases may be completely missing.) Try turning up the IVOL to remedy the problem.

If IVOL is set too high, the recorded phrases may be recorded louder than pre-recorded speech or speech heard while connected to a bridge to another person. The automatic gain control generally prevents this from being a problem, but if recorded speech appears to be to loud, try using a lower IVOL setting.

Finally, subjectivity plays a large role in the effectiveness of a TLP. What sounds appropriate to one person may sound inappropriate to another. The default IVOL and OVOL parameters have been carefully selected to provide appropriate volume levels in the majority of applications. It is strongly recommended that you do not change them based on subjective evaluation. However, the flexibility is provided to tune them to whatever suits the needs of the application at hand.

Transmission Level Plan and Call Bridging

When two incoming calls are bridged together by the VIS, the callers on either end (station set A and station set B) can talk with each other through the VIS. In such a situation, the previously discussed IVOL and OVOL parameters do not apply. Instead, software on the VIS machine (specifically the TSM process) has built in rules for directing the VIS Network Interface cards to insert up to +6 dB gain in either direction of a call bridge connection.

Figures 2-7 through 2-10 depict the rules governing the amount of gain inserted. Recall that the VIS TLP dictates that there be a gain of -6 dB from station-set-to-station-set. Assuming the typical network TLP characteristics for the network facilities (as discussed in the previous section), Figure A-3 through Figure A-6 show the amount of gain (in dB) that is automatically inserted in each direction for each of the four possible call bridging scenarios.

Figure A-3 shows analog-to-analog (T/R-to-T/R) call bridging. Figure A-4 shows digital-to-digital (T1-to-T1) call bridging. Figure A-5 shows analog-to-digital (T/R-to-T1) call bridging. Figure A-6 shows digital-to-analog (T1-to-T/R) call bridging.

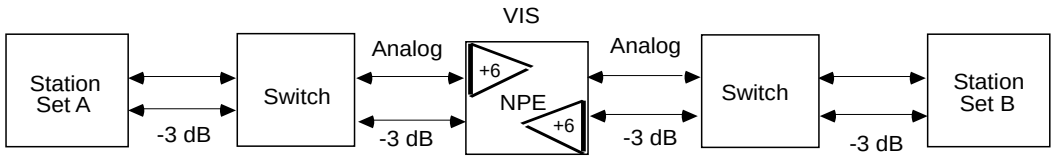


Figure A-3. Analog-to-Analog Call Bridging

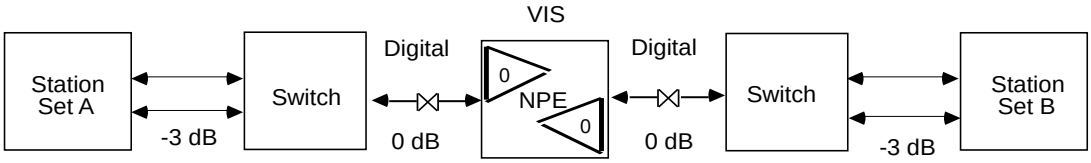


Figure A-4. Digital-to-Digital Call Bridging

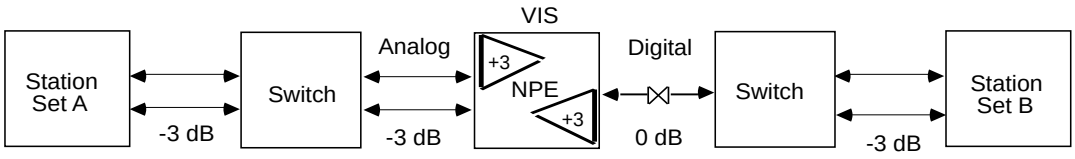


Figure A-5. Analog-to-Digital Call Bridging

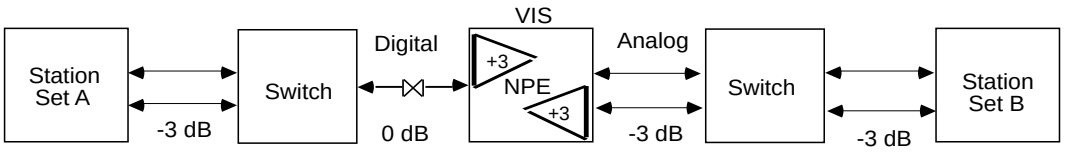


Figure A-6. Digital-to-Analog Call Bridging

Possible Exceptions to the INTUITY CONVERSANT VIS TLP

When a VIS is used as a network adjunct within the AT&T network, some changes to the default TLP settings are recommended to ensure optimal speech volume and clarity. Similar conditions may apply to commercial customers providing voice-response services that are primarily accessed via the long distance network.

NOTE:
Customers should check with their switch and/or network services provider before deviating from the INTUITY CONVERSANT VIS TLP.

In addition to the 6-dB end-to-end loss described above, the FCC requires that the LEC insert a 6-dB loss as signals leave the long distance network. AT&T Truevoice® feature adds up to a gain of 4 dB as low volume level signals leave the AT&T network. This partially compensates for the loss of the 6 dB that the LEC is required to insert.

Within the AT&T network, network recordings and announcements (and operator speech) should be presented at a volume level of -21 dBm0 at the AT&T Point of Presence. If recordings and announcements are recorded at a volume level that is too high, the calling party is likely to hear distortion. This distortion is due to the clipping that occurs when high volume levels exceed the capability of the network to represent the signal. Clipping can occur at -13 dBm0. Excessive volume levels on prerecorded speech is one of the most frequent causes of hearing distortion.

Within the AT&T network, all trunks and bridges should insert zero gain so that the volume level remains as -21 dBm0 throughout the AT&T network.

When an INTUITY CONVERSANT VIS is being used as a network adjunct and digital trunks are used, it is recommended that IVOL and OVOL settings be set to the non-default value of 1000 (for zero gain and that prerecorded speech be recorded at -21 dBm0. By using zero gain, the VIS being used as the network

adjunct may avoid introducing another digital signal transformation that contributes to the distortion heard by users of the network.

When the quality of speech is more important than minimizing space usage (as for most prerecorded announcements and prompts), encode the speech using 64 Kbps PCM rather than 32 Kbps ADPCM.

When the highest quality speech is required, ISDN PRI may provide slightly better sound quality than T1 E&M Robbed-bit signalling (see Chapter 3, “Digital Telephony Interfaces”), where the least significant bits rather than voice data are used for signalling. However, the difference in sound quality is not the only advantage to using ISDN PRI.

The AT&T Truevoice processing inserts a gain of up to 14 dB at low frequencies (around 180 Hz). This is designed to compensate for the normal losses in the analog loop and in telephone handsets. This helps to make low frequency voices sound richer and more like the person is nearby. A 25 Hz (inaudible) tone is used to prevent doubling of the AT&T Truevoice effect in bridges and speech recorded via the long distance network. This tone is lost in an analog bridge or recording made over an analog interface. Fortunately, most of the inserted gain is also lost so there is not a full doubling effect of AT&T Truevoice. When a digital bridge is used or a digital interface is used to make a recording, the 25 Hz tone is preserved along the enhanced signal and the AT&T Truevoice effect is not applied twice. Unfortunately, it may take about a second for the 25 Hz tone to be recognized and for the redundant Truevoice processing to be disabled.

To prevent problems with excessive volume levels from enhanced AT&T Truevoice processing, it is recommended that recordings and announcements be recorded in studio labs (rather than via the network) and that the low frequencies should not be enhanced by the studio.

T/R Switch Integration Issues

Switch integration for T/R circuit cards is done using the Analog Interfaces screen. This screen is described in Chapter 6, “Switch Interface Administration,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. The T/R interface is administered on a system-wide basis, that is, the T/R parameters apply to all T/R circuit cards. To administer the T/R interface, you may specify several parameters or accept the default values.

NOTE:
The IVOL, OVOL, and TDM output gains are system-wide parameters for analog interfaces and can be changed on a per-card basis for digital interfaces. These parameters can be modified via the Switch Interfaces screens as described in Chapter 6, “Switch Interfaces,” of *Intuity* ™ CONVERSANT® System Version 6.0 *Administration*, 585-310-591. Gains can also be overridden on a per-channel basis by an IRAPI application. However, even with IRAPI, the IVOL cannot be overridden for speech recording on a T/R channel. See *Intuity* ™ CONVERSANT® System Version

6.0 *Application Development with Advanced Methods*, 585-310-761, for the IRP_PLAYGAIN and IRP_RECORD_GAIN parameters under IrPARAMETERS(4IRAPI).

All T/R lines originating from the Merlin Legend switch connected to the VIS must be setup in a Merlin Legend calling group as type “Generic VMI.”

Refer to "Introduction to Analog Communications" on page 2-1 of this document for a discussion of the base analog telephony features of the VIS.

Calculating Volume Settings

The next few paragraphs offer a method for calculating the volume settings just described. The same method applies for calculation of incoming text volume (IVOL) and for outgoing text volume (OVOL). The method applies equally to both SP and SSP cards.

Calculation of volume settings for the VIS is very similar to calculation of relative voltage levels. So that volume settings take the form of integers, however, the equation is calculated relative to the (arbitrary) constant of 1000 rather than to a second voltage: $dB = 20\log\frac{Vol}{1000}$.

To calculate a setting where the volume level to be set is known and is expressed in dB, the required setting becomes relative to the inverted log (or antilog) of 10: $Vol = 1000 \times 10^{(dB \div 20)}$

Using this formula, what (for example) would be the setting required to get an OVOL level of -3dB? The formula would look like this: $OVOL = 1000 \times 10^{(-3 \div 20)}$ which becomes: $OVOL = 1000 \times 10^{-0.15}$ or: $OVOL = (1000 \times 0.707) = 707$. The setting would be 707.

The below table sets out the results of this calculation in 3dB increments from -21dB to 21dB.

Table A-2. Loss and Gain Settings

dB Loss	Setting	dB Gain	Setting
0 dB	1000	0 dB	1000
-3 dB	707	3 dB	1412
-6 dB	501	6 dB	1995
-9 dB	354	9 dB	2818
-12 dB	251	12 dB	3981

Table A-2. Loss and Gain Settings

dB Loss	Setting	dB Gain	Setting
-15 dB	177	15 dB	5623
-18 dB	125	18 dB	7943
-21 dB	89	21 dB	11220

Abbreviations

A

AC
Alternating current

ACD
Automatic call distributor

AD
Application dispatch

AD-API
Application dispatch application programming interface

ADPCM
Adaptive differential pulse code modulation

ADU
Asynchronous data unit

AGL
Application generation language

ALERT
System alerter process

ANI
Automatic number identification

API
Application programming interface

ARU
Alarm relay unit

ASAI
Adjunct/Switch Application Interface

ASCII
American Standard Code for Information Interchange

ASI
Analog switch integration

ASR
Advanced Speech Recognition

AYC2C
The signal processor (SP) circuit card

AYC3B
A T1 (digital) circuit card

AYC5B
The IVP6 Tip/Ring (analog) circuit card

AYC6B
The IVP4 Tip/Ring (analog) circuit card.

AYC7
The companion (CMP) circuit card.

AYC9
The Text-to-Speech circuit card

AYC10
The IVC6 Tip/Ring (analog) circuit card

AYC11
A T1 (digital) circuit card

AYC16
The IVP6-IU Tip/Ring (analog) circuit card

AYC21
The E1/T1 (digital) circuit card

AYC26
The IVP6-IA Tip/Ring (analog) circuit card

AYC27
The IVP6-ID Tip/Ring (analog) circuit card

AYC28
The IVP6 Tip/Ring (analog) circuit card

AYC30
The NGTR (analog) circuit card

AYC43
The speech and signal processor (SSP) circuit card

B

BB
Bulletin board

bps
Bits per second

BRDG
Call bridging process

BSC
Binary synchronous communication

C

CCA
Call classification analysis

CDH
Call data handler

CELP
Code Excited Linear Prediction

CGEN
Voice system general message class

CICS
Customer Information Control System

CMP
Companion circuit card

CMS
Call Management System

CO
Central office

CPE
Customer provided equipment or customer premise equipment

CPN
Calling party number

CPT
Call progress tones

CPU
Central processing unit

CSU
Channel service unit

CVS
Converse vector step

D

dB
Decibel

DB
Database

DBC
Database checking process

DBMS
Database management system

DC
Direct current

DCE
Data communications equipment

DCP
Digital communications protocol

DIMM
Dual in-line memory module

DIO
Disk input and output process

DIP
Data interface process

DMA
Direct memory access

DNIS
Dialed number identification service

DPR
Dial Pulse Recognition

DSP
Digital signal processor

DTE
Data terminal equipment

DTMF
Dual tone multifrequency

DTR
Data terminal ready

E

E&M
Ear and Mouth

EBCDIC
Extended Binary Coded Decimal Interexchange Code

ECS
Enterprise Communications Server

EIA
Electronic Industries Association

EISA
Extended Industry Standard Architecture

EMI
Electromagnetic interference

ESD
Electrostatic discharge

ESDI
Extended Serial Data Interface

ESS
Electronic Switching System

ET
Error tracker

EXTA
External alarms feature message class

F

FCC
Federal Communications Commission

FDD
Floppy disk drive

FEP
Front end processor

FFE
Form Filler Plus feature message class

FIFO
First-in-first-out processing order

foos
Facility out-of-service state

FTS
File transfer process message class

G

GEN
PRISM logger and alerter general message class

GSE
Graphical Speech Editor

GUI
Graphical user interface

H

HDD
Hard disk drive

HLLAPI
High Level Language Application Programming Interface

HOST
Host interface process message class

hwoos
Hardware out-of-service state

Hz
Hertz

I

IBM
International Business Machines

ICK
Integrity checking process message class

ID
Identification

IDE
Integrated Disk Electronics

IE
Information element

INIT
Voice system initialization message class

inserv
In-service state

IPC
Interprocess communication

IPC
Intelligent Ports Card (IPC-900)

IPCI
Integrated personal computer interface

IRAPI
INTUITY Response Application Programming Interface

IRQ
Interrupt request

ISA
Industry Standard Architecture

ISDN
Integrated Services Digital Network

ISV
Independent Software Vendor

ITAC
International Technical Assistance Center

IVP4
Integrated Voice Processing card with 4 analog channels

IVP6
Integrated Voice Processing card with 6 analog channels

IVPSS
Integrated Voice Processing System Software

K

Kbps
Kilobites per second

Kbyte
Kilobyte

L

LAN
Local area network

LDB
Local database

LED
Light-emitting diode

LIFO
Last-in-first-out processing order

LN
Load number

LOG
VIS logger process message class

LSE1
Line side E1

LST1
Line side T1

LU
Logical unit

M

manoos
Manually out-of-service state

MAP/100
Multi-Application Platform 100

MAP/100C
Multi-Application Platform 100C

MAP/40
Multi-Application Platform 40

Mbps
Megabits per second

Mbyte
Megabyte

MF
Multifrequency

ms
Millisecond

msec
Millisecond

MHz
Megahertz

MTC
Maintenance process

N

NCP
Network Control Program

NEBS
Network Equipment Building Standards

NEMA
National Electrical Manufacturers Association

netoos
Network out-of-service state

NFAS
Non-Facility Associated Signaling

NFS
Network file sharing

NGTR
Next Generation Tip/Ring

NMVT
Network Management Vector Transport

NM-API
Network Management - Application Programming Interface

nonex
Nonexistent state

NRZ
Non Return to Zero

NRZI
Non Return to Zero Inverted

O

OEM
Original equipment manufacturer

OGA
Operator generated alert

P

P & C
Prompt and Collect

PBX
Private branch exchange

PC
Personal computer

PCB
Printed circuit board

PCM
Pulse code modulation

PEC
Price element code

PRI
Primary rate interface

PSTN
Public switch telephone network

PS&BM
Power supply and battery module

R

RAM
Random access memory

RECOG
Speech recognition feature message class

RDBMS
ORACLE relational database management system

REN
Ringer equivalence number

RFS
Remote file sharing

RM
Resource manager

RMB
Remote maintenance circuit card

RTS
Request to send

S

SBC
Sub-band coding

SCCS
Switching Control Center System

SCSI
small computer system interface

SDLC
Synchronous Data Link Control

SDN
Software Defined Network

SID
Station identification

SIMM
Single in-line memory module

SLIP
Serial Line Interface Protocol

SNA
Systems Network Architecture

SNMP
Simple Network Management Protocol

SP
Signal processor circuit card

SPIP
Signal processor interface process

SPPLIB
Speech processing library

SQL
Structured Query Language

SR
Speech recognition

SYS
UNIX system calls message class

sysgen
System generation

T

TAS
Transaction Assembler Script

TCC
Technology Control Center

TCP/IP
Transmission control protocol/internet protocol

TDM
Time division multiplexing

TE
Terminal emulator

THR
Threshold message class

TKR
Token Ring

TLI
Transport layer interface

TLP
Transmission level plan

T/R
Tip/Ring circuit card

TRIP
Tip/Ring interface process

TSO
Technical Service Organization

TSO
time share operation

TSM
transaction state machine process

TTS
Text-to-Speech

TWIP
T1 interface process

U

UK
United Kingdom

US
United States of America

USOC
Universal service ordering code

UVL
Unified Voice Library

V

VDC
Video display controller

VPC
Voice processing comarketer

VRU
Voice response unit

VROP
Voice response output process

Glossary

Numerics

23B+D
23 bearer (communication) and 1 data (signaling) channel on a T1 PRI circuit card.

30B+D
30 bearer (communication) and 1 data (signaling) channel (plus framing channel 0) on an E1 PRI circuit card.

3270 interface
A link between one or more INTUITY CONVERSANT machines and a host mainframe. In INTUITY CONVERSANT documentation, the 3270 interface specifically means the link between one or more system machines and an IBM host mainframe.

47B+D
47 bearer (communication) and 1 data (signaling) channel on two T1 PRI circuit cards.

4ESS
A large Lucent central office switch used to route calls through the telephone network.

A

adaptive differential pulse code modulation (ADPCM)
A means of encoding analog voice signals into digital signals by adaptively predicting future encoded voice signals. This adaptive modulation method reduces the number of bits required to encode voice. See also “pulse code modulation.”

adjunct products
Products (for example, the Adjunct/Switch Application Interface) that the INTUITY system administers via cut-through access to the inherent management capabilities of the product itself; this is in opposition to the ability of the CONVERSANT system to administer the switch directly.

Adjunct/Switch Application Interface (ASAI)
An optional feature package that provides an Integrated Services Digital Network-based interface between Lucent Technologies PBXs and adjunct processors.

advanced speech recognition (ASR)
A speech recognition ability that allows the system to understand WholeWord and FlexWord® inputs from callers.

affiliate
A business organization that Lucent controls or with which Lucent is in partnership.

alarm relay unit
A unit used in central office telecommunication arrangements that transmits warning indicators from telephone communications equipment (such as an INTUITY CONVERSANT system) to audio.

alerter
A system process that responds to patterns of events logged by the “logdaemon” process.

American Standard Code for Information Interchange (ASCII)
A standard code for data representation that represents alphanumeric characters as binary numbers. The code includes 128 upper- and lowercase letters, numerals, and special characters. Each alphanumeric and special character has an ASCII code (binary) equivalent that is 1 byte long.

analog
An analog signal, such as voice or music, that varies in a continuous manner. An analog signal may be contrasted with a digital signal, which represents only discrete states.

announcement
A message the system plays to the caller to provide information. The caller is not asked to give a response. Compare to “prompt.”

application
The automated transaction (interactions) among the caller, the voice response system, and any databases or host computers required for your business. See also “application script.”

application administration
The component of the INTUITY CONVERSANT system that provides access to the applications currently available on your system and helps you to manage and administer them.

application installation
A two-step process in which the INTUITY CONVERSANT system invokes the TSM script assembler for the specific application name and moves files to the appropriate directories.

application script
The computer program that controls the application (the transaction between the caller and the system). The INTUITY CONVERSANT system provides several methods for creating application scripts, including Graphical Designer, Script Builder, Transaction Assembler Script (TAS) language, and the Intuity Response Application Programming Interface (IRAPI).

application verification
A process in which the INTUITY CONVERSANT system verifies that all the components needed by an application are complete.

asynchronous communication
A method of data transmission in which bits or characters are sent at irregular intervals and spaced by start and stop bits rather than by time. Compare to “synchronous communication.”

asynchronous data unit
An electronic communications device that allows computer systems to communicate over asynchronous lines more than 50 feet (15 m) in length.

automatic call distributor (ACD)
That part of a telephone system that recognizes and answers incoming calls and completes these calls based on a set of instructions contained in a database. The ACD can send the call to an operator or group of operators as soon as the operator has completed a previous call or after the system has played a message to the caller.

automatic number identification (ANI)
A method of identifying the calling party by automatically receiving a string of digits that identifies the calling station of a particular customer.

AYC2C
The signal processor (SP) circuit card.

AYC3B
A T1 (digital) circuit card.

AYC5B
The IVP6 Tip/Ring (analog) circuit card.

AYC6B
The IVP4 Tip/Ring (analog) circuit card.

AYC7
The companion (CMP) circuit card.

AYC9
The Text-to-Speech circuit card.

AYC10
The IVC6 Tip/Ring (analog) circuit card.

AYC11
A T1 (digital) circuit card.

AYC16
The IVP6-IU Tip/Ring (analog) circuit card.

AYC21
The E1/T1 (digital) circuit card.

AYC26
The IVP6-IA Tip/Ring (analog) circuit card.

AYC27
The IVP6-ID Tip/Ring (analog) circuit card.

AYC28
The IVP6 Tip/Ring (analog) circuit card.

AYC30
The NGTR (analog) circuit card.

AYC43
The speech and signal processor (SSP) circuit card.

B

back up
The preservation of the information in a file in a different location, so that the data is not lost in the event of hardware or system failure.

backing up an application
Using a utility that makes an archive copy of a completed application or an interim copy of an application in progress. The back-up copy can be restored to the system if the on-line version is damaged, or if you make revisions and want to go back to the previous version.

barge-in
A capability provided by WholeWord speech recognition and Dial Pulse Recognition (DPR) that allows callers to speak or enter their responses during the prompt and have those responses recognized (similar to the Speak with Interrupt capability). See also “echo cancellation.”

batch file
A file containing one or more lines, each of which is a command executable by the UNIX shell.

binary synchronous communications (BCS)
A character-oriented synchronous link protocol.

blind transfer protocol
A protocol in which a call is completed as soon as the extension is dialed, without having to wait to see if the telephone is busy or if the caller answered.

bridging
The process of connecting one telephone network connection to another over the INTUITY CONVERSANT system TDM bus. Bridging decreases the processing load on the system since an active bridge does not require speech processing, database access, host activity, etc., for the transaction.

bundle
In the context of the Enhanced File Transfer package, this term is used to denote a single file, a group of files (package), or a combination of both.

byte
A unit of storage in the computer. On many systems, a byte is 8 bits (binary digits), which is the equivalent of one character of text.

C

call classification analysis (CCA)
A process that enables application designers to use information available within the system to classify the disposition of originated and transferred calls. Intelligent CCA is provided with the system. Full CCA is an optional feature package.

call data event
A parameter that specifies a list of variables that are appended to a call data record at the end of each call.

call data handler (CHD) process
A software process that accumulates generic call statistics and application events.

called party number
The number dialed by the person making a telephone call. Telephone switching equipment can use this number to selectively route an incoming call to a particular department or agent.

caller
The party who calls for a service, gets connected to the INTUITY CONVERSANT system, and interacts with it. As the INTUITY CONVERSANT system can also make outbound calls for service, the caller can also be the person who responds to those outbound calls.

call progress tones (CPT)
Standard telephony sounds that indicate the status of the call. These sounds include busy, fast busy, ringback, reorder, etc.

card cage
An area within a INTUITY CONVERSANT system platform that contains and secures all of the standard and optional circuit cards used in the system.

cartridge tape drive
A high-capacity data storage/retrieval device that can be used to transfer large amounts of information onto high-density magnetic cartridge tape based on a predetermined format. This tape can be removed from the system and stored as a backup, or used on another system.

caution
An admonishment or advisory statement used in INTUITY CONVERSANT documentation to alert the user to the possibility of a service interruption or a loss of data.

central office (CO)
An office or location in which large telecommunication devices such as telephone switches and network access facilities are maintained. These locations follow strict installation and operation requirements.

central processing unit (CPU)
See "processor."

channel
See "port."

channel associated signaling (CAS)
A type of signaling that can be used on E1 circuit cards. It occurs on channel 16.

circuit card upgrade
A new circuit card that replaces an existing card in the platform. Usually the replacement is an updated version of the original circuit card to replace technology made obsolete by industry trends or a new system release.

cluster controller
A bisynchronous interface that provides a means of handling remote communication processing.

CMP (AYC7)
The companion circuit card to the signal processor (SP).

code excited linear prediction (CELP)
A means of encoding analog voice signals into digital signals that provides excellent quality with use of minimum disk space.

command
An instruction or request the user issues to the system software to make the system perform a particular function. An entire command consists of the command name and options.

CompuLert/SCCS interface
An optional feature that enables remote or console monitoring of error messages generated from the INTUITY CONVERSANT system. CompuLert is a centralized maintenance system for monitoring minicomputers, computer mainframes, etc. The Switching Control Center System (SCCS) is similar to the CompuLert system, but is used to support 4ESS local switching systems.

configuration
The arrangement of the software and hardware of a computer system or network. The INTUITY CONVERSANT system configuration includes either a standard or custom processor, peripheral equipment (for example, printers and modems), and software applications. Configuration also refers to the way the switch network is set up; that is, the types of products that are in the network and how those products communicate.

configuration management
The component of the system that allows you to manage the current configuration of voice channels, host sessions, and database connections, assign scripts to run on specific voice

channels or host sessions, assign functionality to SSP and E1/T1 circuit cards, and perform various maintenance functions.

connect and disconnect (C and D) tones
DTMF tones that inform the system when the attendant has been connected (C) and when the caller has been disconnected (D).

connected digits
A sequence of digits that the system can process as a group, rather than requiring the caller to enter the digits one at a time.

Converse Data Return (conv_data)
A Script Builder action that supports the DEFINITY call vectoring (routing) feature by enabling the switch to retain control of vector processing in the system environment. It supports the DEFINITY "converse" vector command to establish a two-way routing mechanism between the switch and the system to facilitate data passing and return.

controller circuit card
A circuit card used on a computer system that controls its basic functionality and makes the system operational. These circuit cards are used to control magnetic peripherals, video monitors, and basic system communications.

copying an application
A utility in which information from a source application is directed into the destination application.

coresidency
The ability of two products or services to operate and interact with each other on a single hardware platform. An example of this is the use of INTUITY CONVERSANT along with a package from a different vendor on the same system platform.

crash
An interactive utility for examining the operating system core and for determining if system parameters are being exceeded.

custom speech
Unique words or phrases to be used in INTUITY CONVERSANT system voice prompts that Lucent Technologies custom records on a per-customer basis.

custom vocabulary
A specialized package of unique words or phrases created on a per-customer basis and used by WholeWord or FlexWord speech recognition.

Customer Information Control System (CICS)
Part of the operating system that manages resources for running applications (for example, IND$FILE). Note that TSO and CMS provide analogous functionality in other host environments.

D

danger
An admonishment or advisory statement used in INTUITY CONVERSANT documentation to alert the user to the possibility of personal injury or death.

data interface process (DIP)
A software process that communicates with Script Builder applications.

database
A structured set of files, records, or tables.

database field
A field used to extract values from a local database and form the structure upon which a database is built.

database record
The information in a database for a person, product, event, etc. The database record is made up of individual fields for each information item.

database table
A structure, made up of columns and rows, that holds information in a database. Database tables provide a means of storing information that changes too often to "hard-code," or store permanently, in the transaction outline.

debug
The process of locating and correcting errors in computer programs; also referred to as "troubleshooting."

default
The way a computer performs a task in the absence of other instructions.

default owner
The owner of a channel when no process takes ownership of that channel. The default owner holds all idle, in-service channels. In terms of the IRAPI, this is typically the Application Dispatch process.

diagnose
The process of performing diagnostics on a bus or on Tip/Ring, E1/T1, or SSP circuit cards.

dial ahead
The ability to collect and process touch-tone inputs in sequence, even when they are received before the prompts.

dial pulse recognition (DPR)
A method of recognizing caller pulse inputs from a rotary telephone.

dialed number identification service (DNIS)
A service that allows incoming calls to contain information about the telephone number for which it is destined.

dial through
A capability provided by touch-tone and dial pulse recognition that allows callers to enter their responses during the prompt and have those responses recognized (similar to the Speak with Interrupt capability). See also "barge-in" and "echo cancellation".

dictionary
A reference book containing an alphabetical list of words, with information given for each word including meaning, pronunciation, and etymology.

directory
A type of file used to group and organize other files or directories.

display errdata
A command that displays system errors sent to the logger.

dual 3270 links
A feature that provides an additional physical unit (PU) for a cost-effective means of connecting to two host computers. The customer can connect a system to two separate FEPs or to a single FEP shared by one or more host computers. Each link supports a maximum of 32 LUs.

dual tone multi-frequency (DTMF)
A touch-tone sound that is an audio signal including two different frequencies. *DTMF feedback* is the process of the "switch" providing this information to the system. *DTMF muting* is the process of ignoring these tones (which might be simulated by human speech) when they are not needed for the application.

dump space
An area of the disk that is fixed in size and should equal the amount of RAM on the system. The operating system "dumps" an image of core memory when the system crashes. The dump can be fetched after rebooting to help in analyzing the cause of the crash.

E

E1 / T1
Digital telephony interfaces, commonly called *trunks*. E1 is an international standard at 2.048 Mbps. T1 is a North American standard at 1.544 Mbps.

Ear and Mouth (E&M)
A common T1 trunking protocol for connection between two "switches."

echo cancellation
The process of making the channel quiet enough so that the system can hear and recognize WholeWord and dial pulse inputs during the prompt. See also "barge-in."

editor system
A system that allows speech phrases to be displayed and edited by a user. See "Graphical Speech Editor."

Enhanced Basic Speech
Pre-recorded speech available from Lucent Technologies in several languages. Sometimes called "standard speech."

Enhanced File Transfer
A feature that allows the transferring of files automatically between the INTUITY CONVERSANT system and a synchronous host processor on a designated logical unit.

Enhanced Serial Data Interface (ESDI)
A software- and hardware-controlled method used to store data on magnetic peripherals.

Enterprise Communications Server (ECS)
The telephony equipment that connects your business to the telephone network. Sometimes called a "switch."

error message
A message on the screen indicating that something is wrong with a possible suggestion of how to correct it.

Ethernet
A name for a local area network that uses 10BASE5 or 10BASE2 coaxial cable and InterLAN signaling techniques.

event
The notification given to an application when some condition occurs that is generally not encountered in normal operation.

external actions
Specific predefined system tasks that Graphical Designer or Script Builder can call or *invoke* to interact with other products or services. When an external action is invoked, the systems displays a form that provides choices in each field for the application developer to select. Examples are Call_Bridge, Make_Call, SP_Allocate, SR_Prompt, etc.

external functions
Specific predefined (or customer-created) system tasks that can Graphical Designer or Script Builder can call or *invoke* to interact with other products or services. The function allows the application developer to enter the argument(s) for the function to act on (they are not provided in a choices list). Examples are concat, getarg, length, substring, etc.

F

FAX Actions
An optional feature package that allows the system to send fax messages.

feature
A function or capability of a product or an application within the INTUITY CONVERSANT system.

feature package
An optional package that may contain both hardware and software resources to provide additional functionality to a standard system.

feature_tst script package
A standard CONVERSANT software program that allows a user to perform self-tests of critical hardware and software functionality.

field
See “database field.”

file
A collection of data treated as a basic unit of storage.

file transfer
An option that allows you to transfer files interactively or directly to and from UNIX using the file transfer system (FTS).

filename
Alphabetic characters used to identify a particular file.

FlexWord speech recognition
A type of speech recognition based on subword technology that recognizes phonemes or parts of words in a specific language. See also “subword technology.”

Form Filler Plus
An optional feature package that provides the capability for application scripts to record a caller’s responses to prompts for later transcription and review.

Full CCA
A feature package that augments the types of call dispositions that Intelligent CCA can provide.

function key
A key, labeled F1 through F8, on your keyboard to which the INTUITY CONVERSANT software gives special properties for manipulating the user interface.

G

grammar
The inputs that a recognizer can match (identify) from a caller.

Graphical Speech Editor (GSE)
A window-driven, X Windows/Motif based, graphical user interface (GUI) that can be accessed to perform different functions associated with the creation and editing of speech files for applications.

Graphical Designer
An optional software package that provides a graphical interface to assist in development of voice response applications on the INTUITY CONVERSANT system (see also *Script Builder*).

H

hard disk drive
A high-capacity data storage/retrieval device that is located inside a computer platform. A hard disk drive stores data on nonremovable high-density magnetic media based on a predetermined format for retrieval by the system at a later date.

hardware
The physical components of a computer system. The central processing unit, disks, tape, and floppy drives, etc., are all hardware.

Hardware Resource Allocator
A software program that resolves or blocks the allocation of CPU and memory resources for controlling and optional circuit cards.

hardware upgrade
Replacement of one or more fundamental platform hardware components (for example, the CPU or hard disk drive), while the existing platform and other existing optional circuit cards remain.

High Level Language Applications Programming Interface (HLLAPI)
An application programming interface that allows a user to write custom applications that can communicate with a host computer via an API.

host computer
A computer linked to a network to provide a range of services, such as database access and computation. The host computer operates in a time-sharing manner with other computers linked to it via the network.

I

iCk
The system integrity checking process.

idle channel
A channel that either has no owner or is owned by its default owner and is onhook.

IND$FILE
The standard SNA file transfer utility that runs as an application under CICS, TSO, and CMS. IND$FILE is independent of link-level protocols such as BISYNC and SDLC.

independent software vendor (ISV)
A company that has an agreement with Lucent Technologies to develop software to work with INTUITY CONVERSANT to provide additional features required by customers.

indexed table
A table that, unlike a nonindexed table, can be searched via a field name that has been indexed.

industry standard architecture (ISA)
A PC bus standard that allows processors and other circuit cards to communicate with each other.

initialize
To start up the system for the first time.

Integrated Services Digital Network (ISDN)
A network that provides end-to-end digital connectivity to support a wide range of voice and data services.

Integrated Voice Processing (IVP) circuit card
The IVP4 or IVP6 circuit card that provides Tip/Ring connections. The NGTR (AYC30) card also provides the same functions.

intelligent CCA
Monitoring the line after dialing is complete to determine whether a busy, reorder (fast busy), or other failure has been encountered. It also recognizes when the extension is answered or if the extension is not answered after a specified number of rings. The monitoring capabilities are dependent on the network interface circuit card and protocol used

interface
The access point of a system. With respect to the INTUITY CONVERSANT, the interface is designed to provide you with easy access to the software capabilities.

interrupt
The termination of voice and/or telephony functions when some condition occurs.

Intuity Response Application Programming Interface (IRAPI)
A library of commands that provide a standard development interface for voice-telephony applications.

IVC6 circuit card (AYC10)
A Tip/Ring (analog) circuit card with six channels.

IVP4 circuit card (AYC6 or AYC6B)
A Tip/Ring (analog) card with four channels.

IVP6 circuit card (AYC5, AYC5B, or AYC28)
A Tip/Ring (analog) card with six channels.

K

keyboard mapping
In emulation mode, this feature enables the keyboard to send 3270 keyboard codes to the host according to a configuration table set up during installation.

keyword spotting
A capability provided by WholeWord speech recognition that allows the system to recognize a single word in the middle of an entire phrase spoken by a caller in response to a prompt.

L

library states
The state information about channel activities maintained by the IRAPI.

line side E1 (LSE1)
A digital method of interfacing an INTUITY CONVERSANT to a PBX or "switch" using E1-related hardware and software.

line side T1 (LST1)
A digital method of interfacing an INTUITY CONVERSANT to a PBX or "switch" using T1-related hardware and software.

listfile
An ASCII catalog that lists the contents of one or more talkfiles. Each application script is typically associated with a separate listfile. The listfile maps speech phrase strings used by application scripts into speech phrase numbers.

local area network (LAN)
A data communications network in a limited geographical area. The LAN provides communications between computers and peripherals.

local database
A database residing on the INTUITY CONVERSANT system.

logical unit (LU)
A type of SNA Network Addressable Unit.

logdaemon
A UNIX system information and error logging process.

logger
See "logdaemon."

logging on/off
Entering or exiting the INTUITY CONVERSANT system software.

M

magnetic peripherals
Data storage devices that use magnetic media to store information. Such devices include hard disk drives, floppy disk drives, and cartridge tape drives.

main screen
The INTUITY CONVERSANT screen from which you are able to enter either the System Administration or Voice System Administration menu.

maintenance process (MTC)
A software process that runs temporary diagnostics and maintains the state of circuit cards and channels.

masked event
An event that an application can ignore (that is, the application can request not to be informed of the event).

master
A circuit card that provides clock information to the TDM bus.

megabyte
A unit of memory equal to 1,048,576 bytes (1024 x 1024). It is often rounded to one million.

menu
Options presented to a user on a computer screen or with voice prompts.

Microsoft
A manufacturer of software products, primarily for IBM-compatible computers.

mirroring
A method of data backup that allows all of the data transactions to the primary hard disk drive to be copied and maintained on a second identical drive in near real time. If the primary disk drive crashes or becomes disabled, all of the data stored on it (up to 1.2 billion bytes of information) is accessible on the second mirrored disk drive.

MS-DOS
A personal computer disk operating system developed by the Microsoft Corporation.

multifrequency (MF)
Dual tone digit signalling (similar to DTMF), used for trunk addressing between network switches or by network operators.

multithreaded application
A single process/application that controls several channels. Each thread of the application is managed explicitly. Typically this means state information for each thread is maintained and the state of the application on each channel is tracked.

N

NetView
An optional feature package that transmits high-priority (major or critical) messages to the host as operator-generated alerts (OGAs) over the 3270 host link. The NetView Alarm feature package does not require a dedicated LU.

next generation (NGTR) Tip/Ring (AYC30) circuit card
An analog circuit card with six channels.

nonindexed table
A table that can be searched only in a sequential manner and not via a field name.

nonmasked event
An event that must be sent to the application. Generally, an event is nonmaskable if the application would likely encounter state transition errors by trying to it.

null value
An entry containing no value. A field containing a null value is normally displayed as blank and is different from a field containing a value of zero.

O

obsolete hardware
Hardware that is no longer supported on the INTUITY CONVERSANT system.

on-line help
Messages or information that appear on the user's screen when a "function key" (F1 through F8) is pressed.

operator-generated alert (OAG)
A system-monitoring message that is transmitted from the INTUITY CONVERSANT system or other computer system to an IBM host computer and is classified as critical or major.

option
An argument used in a command line to modify program output by modifying the execution of a command. When you do not specify any options, the command executes according to its default options.

ORACLE
A company that produces relational database management software. It is also used as a generic term that identifies a database residing on a local or remote system that is created and maintained using an ORACLE RDBMS product.

P

peripheral (device)
Equipment such as printers or terminals that is in addition to the basic processor.

peripheral component interconnect (PCI)
A newer, higher speed PC bus that is gradually displacing ISA for many components.

permanent process
A process that starts and initializes itself before it is needed by a caller.

phoneme
A single basic sound of a particular spoken language. For example, the English language contains 40 phonemes that represent all basic sounds used with the language. The English word

"one" can be represented with three phonemes, "w" - "uh" - "n." Phonemes vary between languages because of guttural and nasal inflections and syllable constructs.

phrase filtering (screening)
The rejection of unrecognized speech. The WholeWord and FlexWord speech recognition packages can be programmed to reprompt the caller if the INTUITY CONVERSANT system does not recognize a spoken response.

phrase tag
A string of up to 50 characters that identifies the contents of a speech phrase used by an application script.

platform migration
See "platform upgrade."

platform upgrade
The process of replacing the existing platform with a new platform.

pluggable
A term usually used with speech technologies, in particular standard speech, to indicate that a basic algorithmic technique has been implemented to accept one or more sets of parameters that tailors the algorithm to perform in one or more languages.

poll
A message sent from a central controller to an individual station on a multipoint network inviting that station to send if it has any traffic.

polling
A network arrangement whereby a central computer asks each remote location whether it wants to send information. This arrangement enables each user or remote data terminal to transmit and receive information on shared facilities.

port
A connection or link between two devices that allows information to travel to a desired location. See "telephone network connection."

Primary Rate Interface (PRI)
An ISDN term for connections over E1 or T1 facilities that are usually treated as trunks.

private branch exchange (PBX)
A private switching system, either manual or automatic, usually serving an organization, such as a business or government agency, and usually located on the customer's premises.

processor
In INTUITY CONVERSANT system documentation, the computer on which UnixWare and INTUITY CONVERSANT software runs. In general, the part of the computer system that processes the data. Also known as the "central processing unit."

prompt
A message played to a caller that gives the caller a choice of selections in a menu and asks for a response. Compare to "announcement."

prompt and collect (P and C)
A message played to a caller that gives the caller a choice of selections in a menu and asks for a response. The responses is collected and the script progresses based on the caller's response.

pseudo driver
A driver that does not control any hardware.

pulse code modulation (PCM)
A digital modulation method of encoding voice signals into digital signals. See also “adaptive differential pulse code modulation.”

R

record
See “database record.”

recognition type
The type of input the recognizer can understand. Available types include touch-tone, dial pulse, and Advanced Speech Recognition (ASR), which includes WholeWord and FlexWord speech recognition.

recognizer
The part of the system that compares caller input to a grammar in order to correctly match (identify) the caller input.

recovery
The process of using copies of the INTUITY CONVERSANT software to reconstruct files that have been lost or damaged. See also “restore.”

remote database
Information stored on a system other than the INTUITY CONVERSANT system that can be accessed by the INTUITY CONVERSANT system.

remote maintenance circuit card
A INTUITY CONVERSANT circuit card, available with a built-in modem, that allows remote personnel (for example, field support) to access all INTUITY CONVERSANT machines. This card is standard equipment on all new MAP/100 and MAP/40 purchases.

reports administration
The component of INTUITY CONVERSANT system that provides access to system reports, including call classification, call data detail, call data summary, message log, and traffic reports.

restore
The process of recovering lost or damaged files by retrieving them from available back-up tapes or from another disk device. See also “recovery.”

restore application
A utility that replaces a damaged application or restores an older version of an application.

reuse
The concept of using a component from a source system in a target system after a software upgrade or platform migration.

roll back
To cancel changes to a database since the point at which changes were last committed.

rollback segment
A portion of the database that records actions that should be undone under certain circumstances. Rollback segments are used to provide transaction rollback, read consistency, and recovery.

S

screen pop
A method of delivering a screen of information to a telephone operator at the same time a telephone call is delivered. This is accomplished by a complex chain of tasks that include identifying the calling party number, using that information to access a local or remote ORACLE database, and pulling a "form" full of information from the database using an ORACLE database utility package.

script
The set of instructions for the INTUITY CONVERSANT system to follow during a transaction.

Script Builder
An optional software package that provides a menu-oriented interface designed to assist in the development of custom voice response applications on the INTUITY CONVERSANT system.

shared database table
A database table that is used in more than one application.

shared speech
Speech that is a part of more than one application.

shared speech pools
A parameter that allows the user of a voice application to share speech components with other applications.

signal processor (SP) circuit card (AYC2, AYC2B, AYC2C, or AYC9d)
A speech processing circuit card that is an older, lower-capacity version of the speech and signal processor (SSP) circuit card (AYC43).

single inline memory modules (SIMMs)
A method of containing random access memory (RAM) chips on narrow circuit card strips that attach directly to sockets on the CPU circuit card. Multiple SIMMs are sometimes installed on a single CPU circuit card.

single-threaded application
An application that runs on a single voice channel.

slave
A circuit card that depends on the TDM bus for clock information.

small computer system interface (SCSI)
A disk drive control technology in which a single SCSI adapter circuit card plugged into a PC slot is capable of controlling as many as seven different hard disks, optical disks, tape drives, etc.

software
The set or sets of programs that instruct the computer hardware to perform a task or series of tasks — for example, UnixWare software and the INTUITY CONVERSANT system software.

software upgrade
The installation of a new version of software in which the existing platform and circuit cards are retained.

source system
The system from which you are upgrading (that is, your system as it exists *before* you upgrade).

speech and signal processor (SSP) circuit card (AYC43)
The high-performance signal processing circuit card introduced in V6.0 capable of simultaneous support for various speech technologies.

speech energy
The amount of energy in an audio signal. Literally translated, it is the output level of the sound in every phonetic utterance.

speech envelope
The linear representation of voltage on a line. It reflects the sound wave amplitude at different intervals of time. This envelope can be plotted on a graph to represent the oscillation of an audio signal between the positive and negative extremes.

speech file
A file containing an encoded speech phrase.

speech filesystem
A collection of several talkfiles. The filesystem is organized into 16-Kbyte blocks for efficient management and retrieval of talkfiles.

speech modeling
The process of creating WholeWord speech recognition algorithms by collecting thousands of different speech samples of a single word and comparing them all to obtain a statistical average of the word. This average is then used by a WholeWord speech recognition program to recognize a single spoken word.

speech space
An area that contains all digitized speech used for playback in the applications loaded on the system.

speech phrase
A continuous speech segment encoded into a digital string.

speech recognition (SR)
The ability of the system to understand input from callers.

standard speech
The speech package available in several languages containing simple words and phrases produced by Lucent Technologies for use with the INTUITY CONVERSANT system. This package includes digits, numbers, days of the week, and months, each spoken with initial, medial, and falling inflection. The speech is in digitized files stored on the hard disk to be used in voice prompts and messages to the caller. This feature is also called Enhanced Basic Speech.

standard vocabulary
A standard package of simple word speech models provided by Lucent Technologies and used for WholeWord speech recognition. These phrases include the digits “zero” through “nine,” “yes,” “no,” and “oh,” or the equivalent words in a specific local language.

string
A contiguous sequence of characters treated as a unit. Strings are normally bounded by white spaces, tabs, or a character designated as a separator. A string value is a specified group of characters symbolized by a variable.

structured query language (SQL)
A standard data programming language used with data storage and data query applications.

subword technology
A method of speech recognition used in FlexWord recognition that recognizes phonemes or parts of words. Compare to “whole-word technology.”

switch
A software and hardware device that controls and directs voice and data traffic. A customer-based switch is known as a "private branch exchange (PBX).

switch hook
The device at the top of most telephones that is depressed when the handset is resting in the cradle (in other words, is *on hook*). The device is raised when the handset is picked up (in other words, when the telephone is *off hook*).

switch hook flash
A signaling technique in which the signal is originated by momentarily depressing the "switch hook."

switch interface administration
The component of the INTUITY CONVERSANT system that enables you to define the interaction between the INTUITY CONVERSANT system and switches by allowing you to establish and modify switch interface parameters and protocol options for both analog and digital interfaces.

switch network
Two or more interconnected telephone switching systems.

synchronous communication
A method of data transmission in which bits or characters are sent at regular time intervals, rather than being spaced by start and stop bits. Compare to "asynchronous communication."

System 75
An advanced digital switch supporting up to 800 lines that provides voice and data communications for its users.

System 85
An advanced digital switch supporting up to 3000 lines that provides voice and data communications for its users.

system administrator
The person assigned the responsibility of monitoring all INTUITY CONVERSANT software processing, performing daily system operations and preventive maintenance, and troubleshooting errors as required.

system architecture
The manner in which the INTUITY CONVERSANT system software is structured.

system message
An event or alarm generated by either the INTUITY CONVERSANT system or end-user process.

system monitor
A component of the INTUITY CONVERSANT system that tests to verify that each incoming telephone line and its associated Tip/Ring or T1 circuit card is functional. Through the "System Monitor" component, you are able to see displays of the Voice Channel and Host Session Monitors.

T

T1
A digital transmission link with a capacity of 1.544 Mbps.

table
See "database table."

talkfile
An ASCII file that contains the speech phrase tags and phrase tag numbers for all the phrases of a specific application. The speech phrases are organized and stored in groups. Each talkfile can contain up to 65,535 phrases, and the speech filesystem can contain multiple talkfiles.

talkoff
The process of a caller interrupting a prompt, so the prompt message stops playing.

target system
The system to which you are upgrading (that is, your system as you expect it to exist *after* you upgrade).

telephone network connection
The point at which a telephone network connection terminates on an Intuity CONVERSANT system. Supported telephone connections are Tip/Ring, T1, and E1.

terminal emulator
Software that allows a PC or UNIX process to look like a specific type of terminal. In particular, it allows the INTUITY CONVERSANT to temporarily transform itself into a “look alike” of an IBM 3270 terminal. In addition to providing full 3270 functionality, the terminal emulator enables you to transfer files to and from UNIX.

Text-to-Speech (TTS)
An optional feature that allows an application to play US English speech directly from ASCII text by converting that text to synthesized speech. The text can be used for prompts or for text retrieved from a database or host, and can be spoken in an application with prerecorded speech. Text-to-Speech application development is supported through Graphical Designer and Script Builder.

ThickNet
A 10-mm (10BASE5) coaxial cable used to provide interLAN communications.

ThinNet
A 5-mm (10BASE2) coaxial cable used to provide interLAN communications.

time-division multiplex
A method of serving a number of simultaneous channels over a common transmission path by assigning the transmission path sequentially to the channels, with each assignment being for a discrete time interval.

Tip/Ring (T/R)
Analog telecommunications using four-wire media.

token ring
A ring type of local area network that allows any station in the network to communicate with any other station.

trace
A command that can be used to monitor the execution of a script.

traffic
The flow of information or messages through a communications network for voice, data, or audio services.

transaction
The interactions (exchanges) between the caller and the voice response system. A transaction can involve one or more telephone network connections and voice responses from the INTUITY CONVERSANT system. It can also involve one or more of the system optional features, such as speech recognition, 3270 host interface, FAX Actions, etc.

transaction assembler script (TAS)
The computer program code that controls the application operating on the voice response system. The code can be produced from Graphical Designer, Script Builder, or by writing directly in TAS code.

transaction state machine (TSM) process
A multi-channel IRAPI application that runs applications controlled by TAS script code.

transient process
A process that is created dynamically only when needed.

troubleshooting
The process of locating and correcting errors in computer programs. This process is also referred to as debugging.

U

UNIX Operating System
A multiuser, multitasking computer operating system originally developed by Lucent Technologies.

UNIX shell
The command language that provides a user interface to the UNIX operating system.

upgrade scenario
The particular combination of current hardware, software, application and target hardware, software, applications, etc.

usability
A measurement of how easy an application is for callers to use. The measurement is made by making observations and by asking questions. An application should have high usability to be successful.

V

vi editor
A screen editor used to create and change electronic files.

virtual channel
A channel that is not associated with an interface to the telephone network (Tip/Ring, T1, LSE1/LST1, or PRI). Virtual channels are intended to run “data-only” applications which do not interact with callers but may interact with DIPs. Voice or network functions (for example, coding or playing speech, call answer, origination, or transfer) will not work on a virtual channel. Virtual channel applications can be initiated only by a “virtual seizure” request to TSM from a DIP.

vocabulary
A collection of words that INTUITY CONVERSANT is able to recognize using either WholeWord or FlexWord speech recognition.

vocabulary activation
The set of active vocabularies that define the words and wordlists known to the FlexWord recognizer.

vocabulary loading
The process of copying the vocabulary from the system where it was developed and adding it to the target system.

voice channel
A channel that is associated with an interface to the telephone network (Tip/Ring, T1, E1, LSE1/LST1, or PRI). Any Intuity CONVERSANT system application can run on a voice channel. Voice channel applications can be initiated by being assigned to particular voice channels or dialed numbers to handle incoming calls or by a "soft seizure" request to TSM from a DIP or the **soft_szr** command.

voice processing co-marketer
A company licensed to purchase voice processing equipment, such as the INTUITY CONVERSANT system, to market and sell based on their own marketing strategies.

voice response output process (VROP)
A software process that transfers digitized speech between system hardware (for example, Tip/Ring and SSP circuit cards) and data storage devices (for example, hard disk, etc.)

voice response unit (VRU)
A computer connected to a telephone network that can play messages to callers, recognize caller inputs, access and update a databases, and transfer and monitor calls.

voice system administration
The means by which you are able to administer both voice- and nonvoice-related aspects of the system.

W

warning
An admonishment or advisory statement used in INTUITY CONVERSANT documentation to alert the user to the possibility of equipment damage.

WholeWord speech recognition (SR)
An optional feature, available in several languages, based on whole-word technology that can recognize the numbers one through zero, "yes", and "no" (the key words). This feature is reliable, regardless of the individual speaker. This feature can identify the key words when spoken in phrases with other words. A string of key words, called *connected digits*, can be recognized. During the prompt announcement, the caller can speak or use touch tones (or dial pulses, if available). See also "whole-word technology."

whole-word technology
The ability to recognize an entire word, rather than just the phoneme or a part of a word. Compare to "subword technology."

wink signal
An interruption of current to a busy lamp indicating that there is a line on hold.

word
A unique utterance understood by the recognizer.

wordlist
A set of FlexWords that are available for recognition by an application during a Prompt & Collect action step.

word spotting
The ability to search through extraneous speech during a recognition.

Index

Numerics

A

B

C

D

E

F

G

H

I

L

M

N

O

P

R

S

T

V

W