



APX 8000™/MAX® TNT/DSLMAX™

Physical Interface Configuration Guide

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About This Guide

What is in this guide

This guide provides the following instructions for an APX 8000™, MAX TNT®, or DSLTNT™ multiservice access concentrator:

- Basic configuration of your unit
- Configuring shelf controller redundancy (APX 8000 only)
- Configuring Ethernet and modem cards
- Configuring T1, E1, DS3, and other network slot cards
- Configuring the unit in a Signaling System 7 (SS7) network
- Configuring call routing
- Provisioning the switch



Note: This manual describes the full set of features for APX 8000, MAX TNT, and DSLTNT units running True Access™ Operating System (TAOS) software version 8.0.2 or later. Some features might not be available with earlier versions or specialty loads of the software.

This manual hereafter refers to your product as a *TAOS unit* except when referring to features specific to a particular unit.



Warning: Before installing your TAOS unit, be sure to read the safety instructions in the *Access Networks Safety and Compliance Guide*. For information specific to your unit, see the “Safety-Related Electrical, Physical, and Environmental Information” appendix in your unit’s hardware installation guide.

What you should know




This guide is for the person who installs, configures, and maintains a TAOS unit. To configure a unit, you need to understand the following:

- Wide Area Network (WAN) concepts
- Local Area Network (LAN) concepts
- Dial-in LAN connections such as Point-to-Point Protocol (PPP) and Multilink PPP (MP)
- Connection cost management and accounting
- Modems
- Frame Relay
- Asynchronous Transfer Mode (ATM)
- IP routing

- Network security

Documentation conventions

Following are all the special characters and typographical conventions used in this manual:

Convention	Meaning
Monospace text	Represents text that appears on your computer's screen, or that could appear on your computer's screen.
Boldface monospace text	Represents characters that you enter exactly as shown (unless the characters are also in <i>italics</i> —see <i>Italics</i> , below). If you could enter the characters but are not specifically instructed to, they do not appear in boldface.
<i>Italics</i>	Represent variable information. Do not enter the words themselves in the command. Enter the information they represent. In ordinary text, italics are used for titles of publications, for some terms that would otherwise be in quotation marks, and to show emphasis.
[]	Square brackets indicate an optional argument you might add to a command. To include such an argument, type only the information inside the brackets. Do not type the brackets unless they appear in bold type.
	Separates command choices that are mutually exclusive.
>	Points to the next level in the path to a parameter or menu item. The item that follows the angle bracket is one of the options that appears when you select the item that precedes the angle bracket.
Key1-Key2	Represents a combination keystroke. To enter a combination keystroke, press the first key and hold it down while you press one or more other keys. Release all the keys at the same time. (For example, Ctrl-H means hold down the Control key and press the H key.)
Press Enter	Means press the Enter, or Return, key or its equivalent on your computer.
Note:	Introduces important additional information.
 Caution:	Warns that a failure to follow the recommended procedure could result in loss of data or damage to equipment.
 Warning:	Warns that a failure to take appropriate safety precautions could result in physical injury.
 Warning:	Warns of danger of electric shock.

Documentation set

The APX 8000/MAX TNT/DSLNT documentation set consists of the following manuals.

- **Read me first:**
 - *Access Networks Safety and Compliance Guide*
Contains important safety instructions and country-specific compliance information that you must read before installing a TAOS unit.
 - *TAOS Command-Line Interface Guide*
Introduces the TAOS command-line environment and shows how to use the command-line interface effectively. This manual describes keyboard shortcuts and introduces commands, security levels, profile structure, and parameter types.
- **Installation and basic configuration:**
 - *APX 8000 Hardware Installation Guide*
Shows how to install APX 8000 hardware and includes APX 8000 technical specifications.
 - *MAX TNT/DSLNT Hardware Installation Guide*
Shows how to install MAX TNT and DSLNT hardware and includes technical specifications for these units.
 - *APX 8000/MAX TNT/DSLNT Physical Interface Configuration Guide* (this guide)
Shows how to configure the cards installed in a TAOS unit and their line attributes for such functions as framing, signaling, and channel usage. It also describes how calls are routed through the system and includes information about configuring the unit in a Signaling System 7 (SS7) environment. This guide explains shelf controller redundancy for an APX 8000 unit.
- **Configuration:**
 - *APX 8000/MAX TNT/DSLNT ATM Configuration Guide*
Describes how to configure Asynchronous Transfer Mode (ATM) operations on a TAOS unit. This guide explains how to configure physical layer attributes and how to create permanent virtual circuit (PVC) and switched virtual circuit (SVC) ATM interfaces. It includes information about ATM direct and ATM-Frame Relay circuits.
 - *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide*
Describes how to configure Frame Relay operations on a TAOS unit. This guide explains physical layer configuration and restrictions and how to create permanent virtual circuit (PVC) and switched virtual circuit (SVC) interfaces. It includes information about Multilink Frame Relay (MFR) and link management, as well as Frame Relay and Frame Relay direct circuits.
 - *APX 8000/MAX TNT/DSLNT WAN, Routing, and Tunneling Configuration Guide*
Shows how to configure LAN and WAN routing for analog and digital dial-in connections on a TAOS unit. This guide includes information about IP routing, Open Shortest Path First (OSPF) routing, Internet Group Management Protocol (IGMP) routing, multiprotocol routers, Virtual Routers (VRouters), and tunneling protocols.
 - *MultiVoice™ for MAX TNT Configuration Guide*
Shows how to configure the MultiVoice application to run on a MAX TNT unit in both Signaling System 7 (SS7) and H.323 Voice over IP (VoIP) configurations.

- **RADIUS: TAOS RADIUS Guide and Reference**
Describes how to set up a TAOS unit to use the Remote Authentication Dial-In User Service (RADIUS) server and contains a complete reference to RADIUS attributes.
- **Administration and troubleshooting: APX 8000/MAX TNT/DSLNT Administration Guide**
Describes how to administer a TAOS unit, including how to monitor the system and cards, troubleshoot the unit, and configure the unit to use the Simple Network Management Protocol (SNMP).
- **Reference:**
 - *APX 8000/MAX TNT/DSLNT Reference*
An alphabetic reference to all commands, profiles, and parameters supported on TAOS units.
 - *TAOS Glossary*
Defines terms used in documentation for TAOS units.

Performing Basic Configuration

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Introduction to basic configuration

Table 1-1 lists the sections describing the tasks you must perform for the TAOS unit basic configuration. The table includes a brief description of each task and lists the commands and parameters you will use.

For information about more advanced configuration of your TAOS unit, see the following configuration guide:

- *APX 8000/MAX TNT/DSLNT ATM Configuration Guide*
- *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide*
- *APX 8000/MAX TNT/DSLNT WAN, Routing, and Tunneling Configuration Guide*

For information about commands, profiles, and parameters, see the *APX 8000/MAX TNT/DSLNT Reference manual*.

Performing Basic Configuration

Introduction to basic configuration

Table 1-1. Basic TAOS unit configuration tasks

Section	Description of task	Related commands or parameters
“Connecting to a new unit” on page 1-3	Connect the TAOS unit to a terminal or workstation and an Ethernet network.	
“Configuring the shelf-controller IP address on a nonredundant unit” on page 1-4	Specify the date and time for the TAOS unit system clock.	admin> set ip-address
“Setting the system date” on page 1-5	Set the correct date and time with the Date command.	admin> date <i>ymmddhhmm</i>
“Setting the system name” on page 1-5	Specify the name of the TAOS unit. This name is used for authentication.	System profile > Name
“Setting the log level” on page 1-5	Specify the level of event information that the TAOS unit displays at the console.	Log profile > Save-level
“Configuring a default gateway” on page 1-6	Designate a default gateway so that the TAOS unit can forward packets for which it has no route.	IP-Route > gateway-address
“Configuring basic DNS information” on page 1-6	Specify a Domain Name System (DNS) server so that you can use names instead of IP addresses to reach IP hosts.	IP-global profile > Domain-name IP-global profile > DNS-primary-server IP-global profile > DNS-secondary-server
“Pinging the TAOS unit from a local host” on page 1-7	After configuring the TAOS unit with its basic settings, you can use Ping to verify that it is communicating on the network.	Ping
“Recommended basic security measures” on page 1-7	Before making the TAOS unit accessible to users, Lucent recommends that you configure some basic security on the unit.	User > Password Serial > Auto-Logout Serial > User IP-global profile > Must-Accept-Address-Assign IP-global profile> Ignore-ICMP-Redirects SNMP profile

Connecting to a new unit

To communicate with a new TAOS unit, you must assign an IP address to the shelf controller. Once this is done, you can perform further configuration over a LAN using Telnet.

Use the following procedures to connect a new TAOS unit, if you have not already done so, and assign an Ethernet IP address.

New APX 8000 unit

Use the following procedure to initially set up an APX 8000 unit:

- 1 Connect a PC terminal or workstation to the serial port on the shelf controller (see the *APX 8000 Hardware Installation Guide*). If the APX 8000 is equipped with redundant shelf controllers, connect to the serial port on the primary controller.
- 2 Connect an Ethernet cable between the network and the Ethernet port on the shelf controller (see the *APX 8000 Hardware Installation Guide*). If the APX 8000 is equipped with redundant shelf controllers, connect to the Ethernet port on the primary controller.
- 3 Configure an IP address and network mask in the `ip-interface` profile.
 - For an APX 8000 unit with one shelf controller, see “Configuring the shelf-controller IP address on a nonredundant unit” on page 1-4.
 - For an APX 8000 unit with redundant shelf controllers, see “Assigning an Ethernet IP address” on page 2-4.
- 4 Verify that the connection and IP address are correct by pinging any device on the network.
- 5 If redundant controllers are used, set the secondary and soft IP addresses (see “Assigning an Ethernet IP address” on page 2-4 and “Defining the soft IP interface for fault tolerance” on page 2-5 for details).
- 6 Exit the terminal or workstation.
- 7 Telnet from a workstation on the LAN. The system will prompt you for the username and password.

User: **admin**
Password: **Ascend**
- 8 Complete the configuration.

New MAX TNT or DSLTNT unit

Use the following procedure to initially set up a MAX TNT or DSLTNT unit:

- 1 Connect a PC terminal or workstation to the serial port on the shelf controller (see the *MAX TNT/DSLTNT Hardware Installation Guide*). Ensure that the speed is set to 9600 bps.
- 2 Connect an Ethernet cable to the network and to the Ethernet port on the shelf controller (see the *MAX TNT/DSLTNT Hardware Installation Guide*).
- 3 Configure an IP address and network mask in the `ip-interface` profile (see “Configuring the shelf-controller IP address on a nonredundant unit” on page 1-4).

Performing Basic Configuration

Configuring the shelf-controller IP address on a nonredundant unit

- 4 Verify that the connection and the IP address are correct by pinging any device on the network.

```
admin> ping 10.10.10.1  
64 bytes from 10.10.10.1: icmp_seq=0 ttl=255 time=0 ms
```

- 5 Exit the terminal or workstation.
- 6 Telnet to the MAX TNT or DSLTNT using a workstation on the LAN. The system will prompt you for a username and password.

```
User: admin  
Password: Ascend
```

- 7 Complete the configuration.

Configuring the shelf-controller IP address on a nonredundant unit

See “Assigning an Ethernet IP address” on page 2-4 for details about IP address configuration for an APX 8000 unit with redundant shelf controllers.

All TAOS units have an Ethernet port on the shelf controller. This Ethernet port is designed for out-of-band management and light traffic loads. It is not intended to be the primary Ethernet interface for the system. If your unit will be routing heavy Ethernet traffic, use an Ethernet card.

To assign an IP address to the Ethernet interface of the shelf controller on a nonredundant APX 8000 or a MAX TNT or DSLTNT, use the Read and List commands to display the controller’s IP-Interface profile, then set the IP-Address parameter. For example:

```
admin> read ip-interface {{1 controller 1 } 0 }  
IP-INTERFACE/{ { shelf-1 controller 1 } 0 } read  
admin> list  
interface-address* = { { shelf-1 controller 1 } 0 }  
ip-address = 0.0.0.0/0  
2nd-ip-address = 0.0.0.0/0  
rip-mode = routing-off  
..  
..  
admin> set ip-address = 10.2.3.4/24  
admin> write
```

After you assign the unit’s hostname and IP address, you might need to modify the host information on your local Domain Name System (DNS) server to include the TAOS unit.

Setting the system date

If the system date displayed on your screen is incorrect, set the correct date and time with the Date command. For example, to set the date and time to October 22, 2000, 8:50 in the morning:

```
admin> date 0010220850
```

The format for setting the date and time is *ymmddhhmm*. Enter the hour in military (24-hour) time.

Setting the system name

You can assign the TAOS unit a system name of up to 24 characters. Because the system name is used for authenticating connections, keep it relatively simple and use only standard characters.

Here is an example of how to set the TAOS unit system name:

```
admin> read system
SYSTEM read
admin> list
name = " "
system-rmt-mgmt = no
use-trunk-groups = no
idle-logout = 0
parallel-dialing = 5
single-file-incoming = yes
admin> set name = apx01
admin> write
```

Setting the log level

While you are configuring the TAOS unit, you might want to increase the log level to display messages that can help you debug configuration settings. First display the current settings, then enter a new log level.

To display the system-wide event-logging parameters, use the Read and List commands:

```
admin> read log
LOG read
admin> list
save-level = info
save-number = 100
syslog-enabled = no
host = 0.0.0.0
facility = local0
```

To change the log level, specify an option for the Save-Level parameter:

```
admin> set save-level = [none|emergency|alert|critical|error|warning|notice|info|debug]
admin> write
```

If your local network supports a Syslog server, you can configure the server's IP address and the Syslog facility number by setting the Host and Facility parameters in this profile.

Configuring a default gateway

If the TAOS unit does not have a route for the destination address of a packet, it forwards the packet to the default router. Most sites use the default router (such as a GRF® router or a UNIX host running the route daemon) to distribute routing tasks among devices. If you do not configure a default route, the TAOS unit drops packets for which it has no route.

You configure the default route in the IP-Route profile. The name of the default IP-Route profile is always Default, and its destination is always 0.0.0.0.

To configure the default route, first use the Read and List commands to display the default IP-Route profile, and then set the Gateway-Address parameter. For example:

```
admin> read ip-route default
IP-ROUTE/default read
admin> list
name* = default
dest-address = 0.0.0.0/0
gateway-address = 0.0.0.0
metric =1
cost =1
preference = 100
third-party = no
ase-type = type-1
ase-tag = c0:00:00:00
private-route = no
active-route = no
admin> set gateway-address = 10.2.3.17
admin> set active-route=yes
admin> write
IP-ROUTE/default written
```

Configuring basic DNS information

The example in this section uses the domain name *abc.com* and sets the IP address of the primary Domain Name System (DNS) server on the local network. Setting this basic information enables you to access IP hosts by name instead of by IP address.

Here is an example that shows how to configure the DNS information:

```
admin> read ip-global
IP-GLOBAL read
admin> list
```

```
domain-name = ""
dns-primary-server = 0.0.0.0
dns-secondary-server = 0.0.0.0
netbios-primary-ns = 0.0.0.0
netbios-secondary-ns = 0.0.0.0
must-accept-address-assign = no
pool-base-address = [ 0.0.0.0 0.0.0.0 ]
..
..
admin> set domain-name = abc.com
admin> set dns-primary-server = 10.1.2.3
admin> set dns-secondary-server = 10.24.112.57
admin> write -f
```

Pinging the TAOS unit from a local host

After you configure the TAOS unit for IP network access, go to an IP host on the local network and use the Ping command to verify that the unit can communicate on the network. For example:

```
host-1% ping 10.2.3.4
```

In addition, you can verify that the TAOS unit is integrated into your DNS system. For example:

```
host-1% ping apx01
```

Recommended basic security measures

The TAOS unit is shipped from the factory with all its security features set to defaults that enable you to configure and set up the unit without any restrictions. Before you make the TAOS unit generally accessible, you must change the default security settings to protect the configured unit from unauthorized access.

Before bringing the TAOS unit online, Lucent recommends performing the following important security measures:

- “Changing the Admin password” on page 1-8
- “Securing the serial port” on page 1-8
- “Assigning a Telnet password” on page 1-8
- “Requiring acceptance of the pool address” on page 1-9
- “Ignoring ICMP redirects” on page 1-9
- “Disabling directed broadcasts” on page 1-9
- “Configuring SNMP access to the unit” on page 1-10

For additional security measures, see the *APX 8000/MAX TNT/DSLNT WAN, Routing and Tunneling Configuration Guide*.

Changing the Admin password

A user who knows the password to the Admin level can perform any operation on the TAOS unit, including changing the configuration. The Admin password is set to **Ascend** by default. Lucent recommends that you assign a secret password immediately to prevent unauthorized users from gaining access to the unit by means of the default password.

Following is an example of changing the Admin password:

```
default> auth admin
Password: Ascend

admin> read user admin
USER/admin read

admin> set password = secret

admin> write
USER/admin written
```

Note that the Allow-Password permission is set to No in the Admin login. Although this setting protects the unit's passwords, it also prevents the Save command from storing passwords in a configuration file. To save passwords in a configuration file, you can set Allow-Password to Yes in the Admin profile, or you can create another User profile for the purpose of backing up the unit and set Allow-Password to Yes in that profile.

Securing the serial port

By default, when users connect to the serial port on the shelf controller, they are logged in with the Admin User profile. To secure the serial port with a username and password, proceed as follows:

- 1 Read the Serial profile:

```
admin> read serial { 1 17 2 }
```
- 2 Set the User profile to null:

```
admin> set user =
```
- 3 Set Auto-Logout to Yes:

```
admin> set auto-logout = yes
```

This setting automatically logs out the current User profile if the Data Terminal Ready signal (DTR) is lost on the serial port.
- 4 Write the profile:

```
admin> write
```

Now users connecting to the serial port must supply a valid username and password for access to the TAOS unit through the serial port.

Assigning a Telnet password

Lucent recommends that you assign a Telnet password, which can be up to 21 characters in length, to prevent unauthorized Telnet sessions. A user who opens a Telnet session to the TAOS unit is prompted to supply this password.

Following is an example of assigning a Telnet password:

```
admin> read ip-global
IP-GLOBAL read

admin> set telnet-password = SDwiw87

admin> write
IP-GLOBAL written
```

All users attempting to access the TAOS unit via Telnet are prompted for the Telnet password. They are allowed three tries, each with a 60-second time limit, to enter the correct password. If all three tries fail, the connection attempt times out.

Requiring acceptance of the pool address

During PPP negotiation, a caller can reject the IP address offered by the TAOS unit and present its own IP address for consideration. For security reasons, you might want to set the Must-Accept-Address-Assign parameter to Yes to ensure that the TAOS unit terminates such a call:

```
admin> read ip-global
IP-GLOBAL read

admin> set must-accept-address-assign = yes

admin> write
IP-GLOBAL written
```

If you enforce acceptance of the assigned address, the Answer-Defaults profile must enable dynamic assignment, the caller's configured profile must specify dynamic assignment, and the caller's PPP dial-in software must be configured to acquire its IP address dynamically. For more details, see the *APX 8000/MAX TNT/DSLNT WAN, Routing and Tunneling Configuration Guide*.

Ignoring ICMP redirects

The Internet Message Control Protocol (ICMP) was designed to find the most efficient IP route to a destination. ICMP redirect packets are one of the oldest route-discovery methods on the Internet. They are also one of the least secure, because ICMP redirects can be counterfeited to change the way a device routes packets. The following commands configure the TAOS unit to ignore ICMP redirect packets:

```
admin> read ip-global
IP-GLOBAL read

admin> set ignore-icmp-redirects = yes

admin> write
IP-GLOBAL written
```

Disabling directed broadcasts

Denial-of-service attacks known as "smurf" attacks typically use ICMP Echo Request packets with a spoofed source address to direct packets to IP broadcast addresses. These attacks are intended to degrade network performance, possibly to the point that the network becomes unusable.

To prevent the TAOS unit router from being used as an intermediary in this type of denial-of-service attack launched from another network, you must disable the TAOS unit from forwarding the directed broadcasts it receives from another network. The following example shows how to disable directed broadcasts that are not generated locally on all IP interfaces of a TAOS unit with a four-port Ethernet card in shelf 1, slot 12:

```
admin> read ip-int {{1 c 1} 0}
IP-INTERFACE/{ { shelf-1 controller 1 } 0 } read

admin> set directed-broadcast-allowed = no

admin> write
IP-INTERFACE/{ { shelf-1 controller 1 } 0 } written

admin> read ip-int {{1 12 1} 0}
IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 } read

admin> set directed-broadcast-allowed = no

admin> write
IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 } written

admin> read ip-int {{1 12 2} 0}
IP-INTERFACE/{ { shelf-1 slot-12 2 } 0 } read

admin> set directed-broadcast-allowed = no

admin> write
IP-INTERFACE/{ { shelf-1 slot-12 2 } 0 } written

admin> read ip-int {{1 12 3} 0}
IP-INTERFACE/{ { shelf-1 slot-12 3 } 0 } read

admin> set directed-broadcast-allowed = no

admin> write
IP-INTERFACE/{ { shelf-1 slot-12 3 } 0 } written

admin> read ip-int {{1 12 4} 0}
IP-INTERFACE/{ { shelf-1 slot-12 4 } 0 } read

admin> set directed-broadcast-allowed = no

admin> write
IP-INTERFACE/{ { shelf-1 slot-12 4 } 0 } written
```

Configuring SNMP access to the unit

For Simple Network Management Protocol (SNMP) access, an SNMP manager must be running on a host on the local IP network, and the TAOS unit must be able to find that host by means of either a static route or RIP. In addition to these restrictions, the TAOS unit has its own SNMP password security (community strings), which you must set up to protect the TAOS unit from being reconfigured from an unauthorized SNMP station.

Overview of SNMP security

The SNMP profile contains SNMP-readable information about the unit and its SNMP security. There are two levels of security:

- Community strings limit access to the TAOS unit to the community of SNMP managers who know the strings.
- Address security excludes SNMP access unless it is initiated from a specified IP address.

Following are the parameters related to SNMP security:

```
SNMP
  enabled = no
  read-community = public
  read-write-community = write
  enforce-address-security = no
  read-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
  write-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
  contact = ""
  location = ""
  queue-depth = 0
```

Enabling SNMP in the TAOS unit

If you leave the Enabled parameter in the SNMP profile set to No (the default), SNMP utilities cannot access the TAOS unit. The following commands enable SNMP on a unit:

```
admin> read SNMP
SNMP read

admin> set enabled = yes

admin> write
SNMP written
```

Setting community strings

You can specify up to 32 characters as the Read-Write-Community string. The following example changes the default community strings:

```
admin> read snmp
SNMP read

admin> list
enabled = yes
read-community = *****
read-write-community = *****
enforce-address-security = no
read-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
write-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
contact = ""
location = here
queue-depth = 0

admin> set read-community = private

admin> set read-write-community = secret

admin> write
SNMP written
```

Setting up address security

If the Enforce-Address-Security parameter is set to No (its default value), any SNMP manager that presents the correct community name is allowed access. If the parameter is set to Yes, the TAOS unit checks the source IP address of the SNMP manager and allows access only to those IP addresses listed in the Read-Access-Host and Write-Access-Host arrays. Each array can include up to five host addresses.

In the following example, commands enforce address security and specify a trusted address for read and write access:

```
admin> read snmp
SNMP read

admin> list
enabled = no
read-community = public
read-write-community = write
enforce-address-security = no
read-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
write-access-hosts = [ 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 0.0.0.0 ]
contact = ""
location = ""

admin> set enforce-address-security = yes

admin> set read-access 1 = 10.2.3.4

admin> set write-access 2 = 10.2.56.123

admin> write
SNMP written
```

Where to go next

For APX 8000 units with two shelf controllers, proceed to Chapter 2 to configure shelf-controller redundancy. Then proceed to the appropriate chapters to configure slot cards for your unit.

For APX 8000 units with a single shelf controller and for MAX TNT and DSLTNT units, proceed to the appropriate chapters to configure slot cards for your unit.

Configuring Shelf-Controller Redundancy (APX 8000)

2

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Overview of redundancy operations

The APX 8000 can operate with a single shelf controller or with two redundant shelf controllers. When the APX 8000 runs with two shelf controllers, one controller takes on the active role of primary controller while the other performs as the passive, secondary controller that automatically takes over control of the system if the primary controller fails. In an APX 8000 with a single shelf controller, no controller redundancy exists.

In an APX 8000 with two shelf controllers, the primary shelf controller performs all controller operations for the APX 8000:

- Managing the slot cards
- Maintaining a central repository of the unit's configurations (including the current NVRAM configuration)
- Performing call control and processing operations
- Managing all centralized functions, such as SNMP access and communication with a RADIUS server.

In addition, all profiles are modified on the primary controller. When a configuration change is made on the primary controller, the entire configuration is copied to the secondary controller. The secondary controller must be loaded with the same boot and operational code as the primary controller.

Shelf-controller startup and primary election

When an APX 8000 with redundant controllers boots up, each shelf controller passes the power-on self tests (POST) during the boot code loading process. The controllers establish communication with each other over the packet bus, and exchange context information (Redundancy profile and Redundancy-Stats profile information) through the heartbeat protocol. Each controller has its own context (known as Context[1] or Context[2]), which is associated with the controller's serial number. The controllers use the context information to track each other's status.

The controllers next elect the controller that will be primary. The election process is based on a hierarchical list of complex criteria. The first criterion in the list is evaluated and if the criterion is found to be true, one of the controllers is made primary. If the criterion is found to be false, the next criterion in the list is evaluated.

Following is an example of the initial criteria that might be used to designate the primary controller:

- 1 If one controller is missing, the existing (current) controller is made primary.
- 2 If one controller is not communicating, the current controller is made primary.
- 3 If both controllers are communicating, the controllers use the Redundancy profile's Primary-Preference setting to determine which controller is primary.
- 4 If Primary-Preference is set to No-Preference, the controller that last acted as the primary controller is made primary.

If a primary controller is still not determined, additional criterion are evaluated. If all election criteria fail to designate a primary controller, the controller with more resources (for example, more RAM) is made primary.

If the criteria cannot determine which controller is primary, the system selects the right controller (slot 42) to be primary and the left controller (slot 41) becomes the secondary controller.

Once a controller is elected as primary, the primary controller proceeds to load operational code. When the primary is finished loading its code, the secondary controller loads its operational image and gets a copy of the profiles.

Note: Both controllers must load the same boot and operational code version.

Normal operation

During normal operation, the two shelf controllers communicate with each other over the packet bus in a back-and-forth heartbeat, exchanging context information. The status lights on each controller indicate the following activities:

- The heartbeat (HRT) status light on each controller visually indicates that the heartbeat protocol is active by blinking on and off every 4 seconds in a regular pattern that alternates between the two controllers. In a TAOS unit that has only one shelf controller, the HRT status light flashes on for 40 milliseconds every 4 seconds.
- The primary (PRI) status light on each controller is lit if the controller is the primary and is off if the controller is secondary.
- The operational (OPR) status light is lit when the operational code is successfully loaded onto the controller.

The secondary controller does not perform controller operations unless the primary controller resets or fails, or if you change the functionality of the shelf controllers. The secondary controller's main role is to monitor the primary and be ready to take over primary controller functions. The secondary controller maintains the current configuration and the fatal-error history log.

Controller switchover

If the primary controller fails, the secondary controller automatically takes over as primary controller. The *new* primary (old secondary) downs all slot cards and then brings the system back up. All connections are dropped. After the primary shelf controller comes up, the slot cards are reset. The system is now ready to take new calls. Each time a controller is selected as the primary controller, an entry is made in the fatal-error history log.

A switchover, when control passes from the primary controller to the secondary, is initiated by one of the following occurrences:

- The primary controller has a hardware or software problem that causes the module to reset. The secondary is assigned to act as the primary.
- You enter the switchover command, `Redundant-Controller-Switch` at the command line interface, which switches control from the primary to the secondary controller.

APX 8000 slot cards communicate with the primary controller through the packet bus. The primary controller is assigned virtual slot number 43, through which communication with the slot cards occurs. If a switchover occurs, the new primary controller inherits virtual slot number 43.

Log messages

Log messages are issued to notify you of significant events related to shelf controller redundancy. For example, the following cases result in a log message:

- A shelf controller becomes primary.
- A fatal log entry is generated when a shelf controller has a software crash.
- A controller becomes primary when no secondary controller is present.
- The primary controller loses heartbeat communication with the secondary controller.
- The primary controller establishes heartbeat communication with the secondary controller.

Configuring the APX 8000 for shelf-controller redundancy

Setting up the APX 8000 for shelf-controller redundancy includes the following tasks:

- Assigning the system IP address
- Assigning the shelf-controller Ethernet IP address
- Assigning the soft IP address
- Configuring shelf-controller redundancy

You can use the `redundant-controller-switch` command-line interface command to switch primary controller functionality from one controller to the other. See “Switching the primary controller at the command-line interface” on page 2-8 for instructions.

Assigning the system IP address

To configure an APX 8000 that has redundant shelf controllers, you must map system IP settings to the unit's soft IP interface. The soft IP interface is associated with the shelf controller that is currently primary.

Set the IP-Global profile's System-IP-Addr parameter to the address of the soft IP interface. System-IP-Addr must not be set to a particular physical interface, such as the address of a shelf controllers. In a redundant shelf-controller system, the physical address of the primary controller changes according to which controller is currently primary, and the system IP address must be a single, unchanging address that always maps to the current primary controller.

Configuration of the soft IP interface address is described in the section "Defining the soft IP interface for fault tolerance" on page 2-5.

Assigning an Ethernet IP address

An APX 8000 creates an IP interface for the Ethernet port of each shelf controller. The IP-Interface profile index is based on each controller's slot number. The left controller slot on the TAOS unit is number 41, and the right controller slot is 42.

To list the IP interfaces, use the `Dir` command, as follows:

```
admin> dir ip-interface
   6  06/17/1999 03:06:00    { { any-shelf any-slot 0 } 0 }
  19  06/21/1999 23:54:02    { { shelf-1 left-controller 1 } 0 }
  19  06/25/1999 17:45:30    { { shelf-1 right-controller 1 } 0 }
```

The IP interface profile indicated by `{ { shelf-1 left-controller 1 } 0 }` is for the shelf controller in the first controller slot. The IP profile indicated by `{ { shelf-1 right-controller 1 } 0 }` is for the shelf controller in the second controller slot. The IP-Interface profile with the zero index `{ { any-shelf any-slot 0 } 0 }` is reserved for the soft IP interface.

Examples of setting shelf-controller Ethernet IP address

Each shelf controller needs to be assigned an IP address. Following are examples that show how to configure the Ethernet IP addresses.

In the following example, the shelf controller in the left controller slot position (slot 41) is the primary controller. The primary controller is assigned the address 192.168.100.1/24:

```
admin> read ip-interface { { 1 41 1 } 0 }
IP-INTERFACE/{ { shelf-1 left-controller 1 } 0 } read
admin> set ip-address = 192.168.100.1/24
admin> write
IP-INTERFACE/{ { shelf-1 left-controller 1 } 0 } written
```

The following commands assign the address 192.168.100.2/24 to the secondary (right) shelf controller. The commands must be performed on the primary (left) controller.

```
admin> read ip-interface { { 1 42 1 } 0 }
IP-INTERFACE/{ { shelf-1 right-controller 1 } 0 } read
```

```
admin> set ip-address = 192.168.100.2/24
admin> write
IP-INTERFACE/{ { shelf-1 right-controller 1 } 0 } written
```

After you assign IP addresses to the controllers, you can verify that the TAOS unit is a valid IP host on its configured networks by pinging other hosts on those networks, as shown in the following example:

```
admin> ping 192.168.100.56
PING 192.168.100.56: 56 Data bytes
64 bytes from 192.168.100.56: icmp_seq=0 ttl=255 time=0 ms
64 bytes from 192.168.100.56: icmp_seq=1 ttl=255 time=0 ms

--- 192.168.100.56: Ping statistics ---
2 packets transmitted, 2 packets received, 0% packet loss
round-trip min/avg/max = 0/0/0 ms
```

Defining the soft IP interface for fault tolerance

The APX 8000 supports an internal soft IP interface that is always available. It is associated only with the primary controller and is hidden from the secondary controller.

The APX 8000 sets up the soft IP interface after you power on the unit and a controller becomes primary. If a switchover occurs and the secondary controller becomes primary, the soft IP interface is initialized and associated with the new primary controller. The soft IP interface address is reachable as long as one IP interface on the APX 8000 (on an Ethernet card, for example) is operational.

The IP-Interface profile with the zero index is reserved for the soft IP interface. For example, the first line of the following `dir` command output shows the zero index:

```
admin> dir ip-interface
 6  06/17/1999 03:06:00    { { any-shelf any-slot 0 } 0 }
19  06/21/1999 23:54:02    { { shelf-1 left-controller 1 } 0 }
19  06/25/1999 17:45:30    { { shelf-1 right-controller 1 } 0 }
```

If RIP is enabled, the APX 8000 advertises the soft IP interface address as a host route (with a prefix length of /32) using the loopback interface. If RIP is not enabled, routers one hop away from the APX 8000 must have a static route to the soft interface address.

Example of setting the soft IP address

You activate the soft IP interface by entering an IP address for { { any-shelf any-slot 0 } 0 }. The following example shows how to set the soft IP address to 192.168.100.128/24:

```
admin> read ip-interface { 0 0 0 }
IP-INTERFACE/{ { any-shelf any-slot 0 } 0 } read
admin> set ip-addr = 192.168.100.128/24
admin> write
IP-INTERFACE/{ { any-shelf any-slot 0 } 0 } written
```

Configuring shelf-controller redundancy

When setting up shelf-controller redundancy, you might need to configure the following profiles:

- Physical interface profiles (such as IP-Interface, Serial, Ethernet, Ether-Info)
- Redundancy profile

Note: Lucent recommends that you modify profiles on the primary controller only. Modified profiles are sent to the secondary controller.

During profile configuration, the `write` and `delete` command-line interface commands check for permission before allowing you to write or delete any profile. Profile writes or deletes are not allowed on the secondary controller, but you can force implementation of the commands if you use the `-f` command option. When you use `-f`, a warning message alerts you that a profile written on the secondary might be overwritten by a transfer from the primary controller.

Physical interface profiles

The profiles of the physical interfaces, such as IP-Interface, Serial, Ethernet, and Ether-Info, are indexed by each controller's slot number. The left shelf-controller slot is 41, and the right shelf-controller slot is 42.

For example, to read the IP Interface profile for the shelf controller in the left controller slot, enter the following command:

```
admin> read ip-interface { { 1 41 1 } 0 }
```

The APX 8000 has only one shelf, which is identified as `shelf-1`.

Redundancy profile

The Redundancy profile maintains each controller's configuration information (context). The shelf controllers exchange context information during heartbeat communications and use it to track each other's status. The context information for each controller is stored as an array and is identified as `Context[1]` or `Context[2]`.

Configuration of the Redundancy profile primarily involves the following subprofiles and parameters:

- Primary-Preference is a parameter that allows the user to indicate a preference for electing a controller as primary.
- Context is a subprofile that contains context subprofiles for both controllers, `Context[1]` and `Context[2]`.
- Context [N] is a subprofile that contains the context information for an individual controller (`Context[1]` or `Context[2]`).

Note: Configuration of the Redundancy profile parameters must be done only on the primary controller. Profiles written on the secondary controller can be overwritten.

Use the `read` and `write` command-line interface commands to make Redundancy the working profile and list the Redundancy profile contents.

```
admin> read redundancy
REDUNDANCY read

admin> list
[in REDUNDANCY]
context = [ { } { } ]
primary-preference = no-preference
```

The following example shows how you can configure the Primary-Preference parameter to indicate a preference for the controller in the right shelf-controller slot to be elected primary:

```
admin> read redundancy
REDUNDANCY read

admin> set primary-preference = right-controller-preferred

admin> write
REDUNDANCY written
```

Note: Primary-Preference settings remain in effect after a reboot. For example, if the left controller is configured with a particular setting, after a reboot the left controller still retains that setting.

The Redundancy-Stats profile contains system-maintained statistical information about each controller. The statistical information for each controller is located in Context-Stats[1] or Context-Stats[2].

The following example shows how you can view the contents of the Redundancy-Stats profile:

```
admin> read redundancy-stats
REDUNDANCY-STATS read

admin> list
[in REDUNDANCY-STATS]
context-stats = [ { monitoring secondary defer-to-running-primary
no-function+

admin> list context 1
[in REDUNDANCY-STATS:context-stats[1]]
state = monitoring
function = secondary
select-reason = defer-to-running-primary
prior-function = no-function
last-reboot = crash
fan = { 317834728 }

admin> list context 2
[in REDUNDANCY-STATS:context-stats[2]]
state = monitoring
function = primary
select-reason = communication-loss
prior-function = no-function
last-reboot = crash
fan = { 317838764 }
```

The Redundancy and Redundancy-Stats profiles are visible through SNMP.

Refer to the *APX 8000/MAX TNT/DSL/TNT Reference* for additional information about the Redundancy and Redundancy-Stats profiles.

Switching the primary controller at the command-line interface

You can manually switch primary shelf-controller functionality to the secondary controller by entering the command `redundant-controller-switch` at the command-line interface. This command causes the primary controller to give up bus (slot card) ownership and allow the other controller to become primary. The switchover to the secondary controller occurs only if the secondary controller is present. After the bus is released, the old primary shelf controller reboots and assumes the role of secondary controller.

Switchover takes place only if the following conditions are met:

- The secondary controller is present.
- The primary controller currently controls the bus.
- The secondary controller requests control of the bus, which is the normal operating state of the secondary controller. The secondary controller is ready to automatically gain bus ownership whenever the primary releases its ownership.

After you use the `redundant-controller-switch` command, a prompt appears that asks for confirmation of your request. To switch primary controller functionality to the secondary controller without being prompted for confirmation, use the `-f` command option, as follows:

```
admin> redundant-controller-switch -f
```

When the command is entered on the primary controller, controller functionality is switched to the secondary controller. When the switchover command is entered on the secondary controller, no switchover occurs.

If the switchover command is entered on the primary when the secondary is not requesting control of the bus, no switchover occurs:

```
admin> redundant-controller-switch
The remote controller is not requesting the bus,
it cannot become PRIMARY!
```

If the switchover command is entered on the primary controller when only one controller is present, a notice is displayed:

```
admin> redundant-controller-switch
There is no remote controller!
```

Resetting shelf controllers and clearing controller NVRAM

The shelf controllers can be reset from the command line with the `reset` command. The controller's NVRAM can be cleared from the command line with the `nvr` command. The use of these commands is described in this section.

Refer to the *APX 8000/MAX TNT/DSLNT Reference* for additional information on the `reset` and `nvr` command-line interface commands and command options.

Resetting the controllers

The `reset` command resets one or both APX 8000 redundant shelf controllers. When you reset the unit, it restarts, and all active connections are terminated. All users are logged out, and

the default security level is reactivated. In addition, a system reset can cause a WAN line to temporarily be shut down due to momentary loss of signaling or framing information. After a reset, the unit runs power-on self-tests (POST).

The `reset -r` command resets the secondary controller or both controllers. The `reset -f` command resets the controller where the command is invoked. When the primary controller is reset, the secondary controller automatically takes over control and becomes primary.

Following is an example of what you enter to reset both controllers:

```
admin> reset -r b
```

Clearing NVRAM

The `nvr` command clears NVRAM and resets one or both APX 8000 redundant shelf controllers.

The `nvr -r` command clears NVRAM and resets the secondary controller or both controllers. The `-f`, `-t`, `-u`, and `-c` command options apply to the controller where the command is invoked. When the `nvr` command is performed on the primary controller and NVRAM is cleared and the controller reset, the secondary controller automatically takes over control and becomes primary.

Enter the following command to clear NVRAM and reset the secondary shelf controller:

```
admin> nvr-r s
```

Enter the following command to clear NVRAM and reboot both shelf controllers:

```
admin> nvr-r b
```

Obtaining status information about redundant shelf controllers

You can use the following methods to obtain information about the redundant shelf controllers:

- The command-line interface `uptime` command indicates the length of time the controllers have been operational.
- The command-line interface `show` command provides status information about the redundant shelf controllers.
- The Trap profile parameter Secondary-Controller-State-Change-Enabled allows a trap to be sent to the NavisAccess™ manager whenever the secondary controller goes in or out of service.

Following are descriptions of these methods.

Viewing controller up time

The `uptime` command reports the length of time the primary controller has been operational. It also indicates the time elapsed since the secondary controller started communications with the primary. If a controller reboots or if communication between the two controllers is

disrupted and then reestablished, the `uptime` command reports the time elapsed since the secondary controller reestablished communications with the primary.

The `uptime` command does *not* report the version number of code used by the controllers, but instead reports the primary or secondary status of each controller. The code version is obtained with the `version` command.

The following example shows the `uptime` command entered on the primary controller. The `-a` option displays the up time for all slot cards.

```
admin> uptime -a
06:28:41
{ shelf-1 slot-3 }      8t1-card      0 days 00:08:53  8.0
{ shelf-1 slot-12 }     hdlc2-card  0 days 00:08:53  8.0
{ shelf-1 slot-16 }     csmx-card   0 days 00:08:53  8.0
{ shelf-1 slot-19 }     hdlc2-card  0 days 00:08:53  8.0
{ shelf-1 slot-23 }     csmx-card   0 days 00:08:53  8.0
{ shelf-1 slot-32 }     hdlc2-card  0 days 00:08:53  8.0
{ shelf-1 slot-34 }     4ether2-card 0 days 00:08:53  8.0
{ shelf-1 left-controller } [...] 0 days 00:40:37 ( SECONDARY )
{ shelf-1 right-controller } [...] 0 days 00:41:21 ( PRIMARY )
```

The following example shows the `uptime` command entered on the secondary controller:

```
admin> uptime -a
06:28:26
{ shelf-1 left-controller } [...] 0 days 00:40:37 ( SECONDARY )
{ shelf-1 right-controller } [...] 0 days 00:41:21 ( PRIMARY )
```

Viewing controller status

The `show` command reports the communications status of the primary and secondary controllers and indicates which controller (left or right) is the primary and secondary shelf controller.

When the `show` command is entered on either the primary or secondary shelf controller, UP is reported for the other controller's status if the current controller is able to communicate with the other controller. DOWN is displayed if the other controller is present but not communicating with the current controller. If the other controller is not present, the status of that controller is reported as ABSENT with the `show -a` command.

The following example displays the `show` command entered on the primary controller, when the right controller is primary:

```
admin> show
Controller { right-controller } ( PRIMARY ):
{ left-controller }      UP      ( SECONDARY )
{ shelf-1 slot-1 0 }     DOWN    ether3-card
{ shelf-1 slot-3 0 }     UP      8t1-card
{ shelf-1 slot-12 0 }    UP      hdlc2-card
{ shelf-1 slot-16 0 }    UP      csmx-card
{ shelf-1 slot-19 0 }    UP      hdlc2-card
{ shelf-1 slot-23 0 }    UP      csmx-card
{ shelf-1 slot-32 0 }    UP      hdlc2-card
{ shelf-1 slot-34 0 }    UP      4ether2-card
```

The following example displays the `show` command entered on the secondary controller when the right controller is primary:

```
admin> show
Controller { left-controller } ( SECONDARY ):
    { right-controller )          UP          ( PRIMARY )
```

Setting up a trap to monitor the secondary controller

In the Trap profile, you can configure the `Secondary-Controller-State-Change-Enabled` parameter to send a trap to the NavisAccess manager whenever the secondary controller goes in or out of service. When the parameter is set to `yes`, a trap is sent when the secondary controller goes in or out of service. When the parameter is set to `no`, no trap is sent.

Use the `read` and `list` commands to make Trap the working profile and list its contents. Use the `set` command to modify the settings in the profile.

The following example shows how to set the parameter to not send a trap to the NavisAccess manager:

```
admin> set secondary-controller-state-change-enabled=no
```

Clearing the fatal-error history log

The `clr-history` command clears the fatal-error history log. In systems with redundant shelf controllers, the `clr-history` command-line interface command is intended only for use on the primary controller. The fatal-error log cannot be cleared on the secondary controller, unless you force implementation of the command by using the `-f` command option.

When `clr-history -f` is used on the secondary controller, a warning message appears to alert you that the cleared log can still be overwritten during transfer of information from the primary controller during heartbeat communications.

Configuring the Thermal Profile for Fan Tray Operations (APX 8000)

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Overview of the Thermal profile for fan tray operations

On the APX 8000, an on-board digital temperature chip on the shelf controller and a temperature device at the intake end of the fan controller are used to measure incoming ambient air temperature. The APX 8000 fan tray is capable of running at different speeds, and of adjusting as needed to dissipate system heat or reduce unnecessary fan noise.

You control fan tray operations by configuring the Thermal profile. Following are the relevant settings, shown with default values:

```
[in THERMAL]
fantray-lownoise-rpm = 2500
operation-mode = full-speed-only
low-temperature-trigger = 34
high-temperature-trigger = 40
alarm-temperature-trigger = 55
```

Parameter	Specifies
Fantray-Lownoise-RPM	Number of revolutions per minute (RPM) of the fan tray when the low noise speed has been selected. Valid values range from 2000 to 3000, with a default of 2500.

Configuring the Thermal Profile for Fan Tray Operations (APX 8000)

Overview of the Thermal profile for fan tray operations

Parameter	Specifies
Operation-Mode	Mode of operation in which the fan tray runs. When the parameter is set to <code>full-speed-only</code> , the fans in the fan tray operate at full speed at all times. (This is the default mode.) When set to <code>lownoise-speed-only</code> , the fans operate at the low noise speed (as specified in the Fantray-Lownoise-RPM setting) at all times. When the parameter is set to <code>auto-regulation</code> mode, the fan speeds are controlled dynamically on the basis of temperature. In <code>auto-regulation</code> mode, the fans run at low noise speed when the system starts up. The system monitors the unit temperature, and when it reaches a high-temperature threshold (as specified in the High-Temperature-Trigger setting), it switches the fans to full speed and logs a message. When the unit temperature falls below the low-temperature threshold (as specified in the Low-Temperature-Trigger setting), the system switches the fans back to low noise speed.
Low-Temperature-Trigger	Low-temperature threshold setting, from 0 to 60 degrees Celsius (32 to 140 degrees Fahrenheit). If the fan tray is in <code>auto-regulation</code> mode and this threshold is crossed, the system switches the fans to low noise speed and logs a message. If you specify a higher value than the High-Temperature-Trigger setting, the system displays an error message when you attempt to write the profile.
High-Temperature-Trigger	High-temperature threshold setting, from 0 to 60 degrees Celsius (32 to 140 degrees Fahrenheit). If the fan tray is in <code>auto-regulation</code> mode and this threshold is crossed, the system switches the fans to full speed and logs a message. If you specify a lower value than the Low-Temperature-Trigger setting, the system displays an error message when you attempt to write the profile.
Alarm-Temperature-Trigger	Temperature threshold setting, from 0 to 60 degrees Celsius (32 to 140 degrees Fahrenheit). If this threshold is crossed, the system generates an Alarm event, the Alarm Relay on the shelf controller is turned on, and the Alarm status light on the front panel of the fan tray illuminates.

Example of configuring thermal controls

The commands in the following example show how to configure the fan tray to run the fans at 2500 RPM until the unit reaches a temperature of 37 degrees Celsius (98.6 degrees Fahrenheit), at which time the system switches the fans to full speed and maintains that setting until the unit temperature drops below 30 degrees Celsius (86 degrees Fahrenheit). If the system ever reaches a temperature of 50 degrees Celsius (122 degrees Fahrenheit), the system triggers alarms.

```
admin> read thermal
THERMAL read

admin> set operation-mode = auto-regulation
```

```
admin> write
THERMAL written

admin> list
[in THERMAL]
fantray-lownoise-rpm = 2500
operation-mode = auto-regulation
low-temperature-trigger = 30
high-temperature-trigger = 37
alarm-temperature-trigger = 50
```

Related log messages

When the fan tray is in auto-regulation mode, the system can generate the following Info log messages to indicate that the system has switched the fans from low noise to full speed, or vice versa:

```
LOG info, Shelf 1, Slot 42, Time: 10:31:39--
Fantray now running in lownoise-mode (30 C)

LOG info, Shelf 1, Slot 42, Time: 10:34:40--
Fantray now running at full speed (37 C)
```

If you modify the fan operation mode setting in the Thermal profile, the system generates an Info log message such as the following:

```
LOG info, Shelf 1, Slot 42, Time: 11:06:44--
Fantray set to run in Auto-regulation mode
```

Thermal alarms

When the temperature of the system reaches the Alarm-temperature-trigger threshold specified in the Thermal profile, the system is in an alarm state. When this happens, the following events occur:

- The system generates an Error log message such as the following:

```
LOG error, Shelf 1, Slot 42, Time: 11:10:23--
Temperature Alarm triggered (50 C)
```
- The Alarm relay on the shelf controller is enabled. This turns on whatever signal is connected to the Alarm relay on the shelf controller.
- The Alarm status light in the fan tray front panel turns ON.

When the temperature falls back 2 degrees Celsius below the Alarm temperature trigger threshold, the alarm state is cleared and the following events occur:

- The system generates a Warning log message such as the following:

```
LOG warning, Shelf 1, Slot 42, Time: 11:11:33--
Temperature Alarm cleared (48 C)
```
- The Alarm relay on the shelf controller is disabled.
- The Alarm status light in the fan tray front panel turns OFF.

The 2-degree temperature cushion retards system response slightly so that the Alarm state is not triggered repeatedly around a threshold.

Thermal status reporting

A power-on self test (POST) is run on the fan tray of the APX 8000 during the BOOT loader and during the operational load. If the fan tray POST fails, the POST failure status light (amber) flashes 10 times and the following log message is generated:

```
LOG emergency, Shelf 1, Slot 42, Time: 15:23:57--
post failed, type (10)
```

In addition, two new commands are supported for displaying information about the fan tray and the unit's thermal status. Both commands, along with the automatic fan tray speed regulation, are available on both shelf controllers in a redundant system.

Fanstatus command

The `fanstatus` command displays fan tray status information such as the fan revolutions per minute (RPM), status (OK or BAD), and the unit's ambient temperature. Note that the current fan mode can be displayed as either full speed or low noise. For example, the following output shows the fan mode set to full speed with an ambient temperature of 33 degrees Celsius (91.4 degrees Fahrenheit):

```
admin> fanstatus
```

```
APX8000 Fantray status
Fantray ambient temperature: 33 C
Current fan mode: Full-speed
```

Fan #	RPM	Status
=====		
1	3367	GOOD
2	3214	GOOD
3	3075	GOOD
4	3075	GOOD
5	3214	GOOD
6	3289	GOOD

The following command output shows the fan mode set to low noise with an ambient temperature of 27 degrees Celsius (80.6 degrees Fahrenheit):

```
admin> fanstatus
```

```
APX8000 Fantray status
Fantray ambient temperature: 27 C
Current fan mode: Low-noise
```

Fan #	RPM	Status
=====		
1	1992	GOOD
2	2050	GOOD
3	1992	GOOD
4	2020	GOOD
5	2050	GOOD
6	2020	GOOD

Thermalstatus command

The `thermalstatus` command displays a number of temperature-related values to show the overall thermal status of the unit. For example, it displays:

- Ambient temperature at fan tray intake.
- Shelf controller temperature.
- High, Low, and Alarm temperature thresholds.
- Slot card temperature for slot cards that support temperature reporting. Currently, no slot cards support thermal information reporting.
- Power supply thermal status, and whether the power supplies are in an overheated state.
- Fan tray status, including the fan tray operational mode, number of revolutions per minute at low-noise speed, and current revolutions per minute of each fan.

For example:

```
admin> thermalstatus
```

System Thermal status

```
Ambient temperature at intake : 27 C (80 F)
Shelf controller temperature   : 35 C (95 F)
High temperature threshold    : 36 C (96 F)
Low temperature threshold     : 32 C (89 F)
Alarm temperature threshold    : 38 C (100 F)
```

Slot cards:

```
(no slot cards contain thermal information)
```

Power supply thermal status

Power Supply #	Temp
=====	
A	OK
B	OK
C	n/a
D	OK

Fantray status

```
Fan operational mode: auto-regulation
Low-noise RPM:       2000
Current fan mode:    Full-speed
```

Fan #	RPM	Status
=====		
1	3289	GOOD
2	3214	GOOD
3	3075	GOOD
4	3143	GOOD
5	3214	GOOD
6	3289	GOOD

Configuring Ethernet Cards

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Introduction to Ethernet slot cards

This chapter explains how to install and configure the Ethernet slot cards. For information about configuring IP routing, see the *APX 8000/MAX TNT DSLTNT WAN, Routing and Tunneling Configuration Guide*.

The following Ethernet slot cards are available for the platforms indicated:

- 10/100Mbps Ethernet-2 card with three 10Mbps ports and one 100Mbps port—APX 8000, MAX TNT and DSLTNT units
- 10/100Mbps Ethernet-3 card with a single 100Mbps port—APX 8000 and MAX TNT units

Full-duplex 10/100Mbps Ethernet-2 slot card

The Ethernet-2 card has three 10BaseT ports and one full-duplex 100BaseT port. If you are replacing an older Ethernet card with the new Ethernet-2 card, you must create new Ethernet profiles for the Ethernet-2 card. For details, see “Upgrading to the Ethernet-2 and Ethernet-3 slot cards” on page 4-2.

Full-duplex 10/100Mbps Ethernet-3 slot card

The Ethernet-3 slot card has one full-duplex 10/100Mbps port that is designed to have a high packet-per-second throughput to support Voice over IP (VoIP). The Ethernet-3 card autosenses 10Mbps or 100Mbps but does not support autonegotiation, in which Ethernet devices negotiate a common speed and duplex mode.

Upgrading to the Ethernet-2 and Ethernet-3 slot cards

To upgrade from an existing 10Mbps or 10/100Mbps Ethernet card to an Ethernet-2 or Ethernet-3 slot card, proceed as in the following example:

- 1 Remove the existing Ethernet slot card.
- 2 Enter the Slot command with the `-r` option to remove the existing Ethernet profiles. For example, if the Ethernet card was in slot 1:

```
admin> slot -r 1
slot 1 removed
```
- 3 Install the Ethernet-2 or Ethernet-3 slot card.
- 4 Configure Ethernet profiles for the new card as explained in the following sections of this chapter.

Overview of Ethernet configuration

The Ethernet slot cards provide multiport Ethernet routing capabilities. The configuration of each port on an Ethernet slot card is identical to the configuration of the Ethernet port on the shelf controller. (For complete information about configuring the Ethernet ports for routing, see the *APX 8000/MAX TNT/DSLNT WAN, Routing, and Tunneling Configuration Guide*.)

All TAOS units have an Ethernet port on the shelf controller. This Ethernet port is designed for out-of-band management and light traffic loads. It is not intended to be the primary Ethernet interface for the system. If your TAOS unit will be routing heavy Ethernet traffic, use an Ethernet card.

Understanding the Ethernet-related profiles

The APX 8000 creates the following profiles when it detects an Ethernet port:

- Ethernet profile
- IP-Interface profile
- SNMP profiles (Admin-State and a Device-State profile)

For an explanation of SNMP profiles, see the *APX 8000/MAX TNT/DSLNT Administration Guide*.

Ethernet profile

TAOS creates a default Ethernet profile for each Ethernet port it detects, including the shelf controller. The Ethernet profile specifies the link-layer configuration for the port.

For example, if an Ethernet-2 card installed in slot 4, you might see a screen similar to the following:

```
admin> dir ethernet
5  08/06/1998 17:03:48 { shelf-1 controller 1 }
5  08/06/1998 17:11:46 { shelf-2 slot-4 1 }
5  08/06/1998 17:11:46 { shelf-2 slot-4 2 }
```

```
5  08/06/1998 17:11:46  { shelf-2 slot-4 3 }
5  08/06/1998 17:11:46  { shelf-2 slot-4 4 }
```

If the 10/100 Mbps Ethernet-2 card is installed, the 100Mbps Ethernet port is displayed as port 4.

IP-Interface profile

TAOS creates a default IP-Interface profile for each Ethernet port it detects, including the shelf controller. You can create multiple IP interfaces for each physical Ethernet port, but the default IP-Interface profile must have an IP address, or the other IP-Interface profiles for the same port will not function. For information about configuring IP-Interface profiles, see the *APX 8000/MAX TNT/DSLNT WAN, Routing, and Tunneling Configuration Guide*.

Configuring duplex mode on the 100Mbps Ethernet port

The Duplex-Mode parameter in the Ethernet profile allows you to set the physical Ethernet interface of the 100BaseT port on the Ethernet-2 or Ethernet-3 card to full-duplex or half-duplex mode. Full-duplex mode (the default) provides increased throughput, but half-duplex mode enables the unit to operate with older equipment that does not support full duplex.

The following example sets the port to half-duplex mode:

```
admin> read ethernet { 1 7 4 }
ETHERNET/{ shelf-1 slot-7 4 } read

admin> list
[in ETHERNET/{ shelf-1 slot-7 4 }]
interface-address* = { shelf-1 slot-7 4 }
link-state-enabled = no
enabled = yes
ether-if-type = utp
bridging-enabled = no
filter-name = ""
duplex-mode = full-duplex

admin> set duplex-mode = half

admin> write
ETHERNET/{ shelf-1 slot-7 4 } written
```


Configuring Series56 II and III Modem and Hybrid Access Cards

Overview of configuring modem cards	5-1
Specifying modem negotiation settings	5-2
Specifying modem modulation for Series56 II and III modem cards	5-3
Configuring an additional AT answer string for modem calls	5-3
Series56 II and III Call-Route profiles	5-4
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Hybrid Access card implementation	5-5

Typically, Series56 II™ and Series56 III Digital Modem slot cards do not require any configuration. Depending on your network, situations might require you to change the way the modems operate. This chapter describes how to modify modem configuration to accommodate your networking environment. The chapter also provides some guidelines for use of Hybrid Access (HDLC) cards.

Note: Modem cards are not supported on DSLTNT units, but DSLTNT units support Hybrid Access cards.

Overview of configuring modem cards

When you make a change to the modem configuration, the change applies to all the modems in the APX 8000 or MAX TNT unit. You configure modems in the Terminal-Server profile.

Table 5-1 lists common tasks you might have to perform to customize modem configurations, the sections describing those tasks, and the associated parameters.

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSLTNT Reference*.

Table 5-1. Modem configuration tasks

Description of task	Section	Associated parameters
Some analog modem calls might require changes to the digital modem's default behavior to successfully complete negotiation.	"Specifying modem negotiation settings" on page 5-2	V42/MNP Max-Baud-Rate Modem-Transmit-Level Cell-Mode-First Cell-Level 7-Even
You might need to change the modulation of Series56 II and III modems from the default of V.90. For example, in some cases, V.32 and V.34 modems do not successfully complete modem training after reception of the V.8bis tone from the APX 8000 and MAX TNT unit Series56 II and III modems. Configuring V.34 modulation can help this problem.	"Specifying modem modulation for Series56 II and III modem cards" on page 5-3	Modem-Mod
You might need to modify the AT answer strings that the APX 8000 and MAX TNT unit sends to its modems. You can do this by specifying an extra answer string in the command line interfaces.	"Configuring an additional AT answer string for modem calls" on page 5-3	AT-Answer-String
Because the Series56 II and III slot cards can terminate both modem and HDLC calls, the APX 8000 and MAX TNT unit creates two call route profiles for each channel on the card: one for a digital call and one for a modem call.	"Series56 II and III Call-Route profiles" on page 5-4	N/A

Specifying modem negotiation settings

Calls from analog modems are directed first to the digital modems, where the connection must be negotiated before being directed to by the terminal-server software. Options in the Terminal-Server > Modem-Configuration subprofile allow you to modify the way the digital modems negotiate a connection.

To specify changes in how the negotiation occurs:

- 1 Read the Terminal-Server profile into the editing buffer:

```
admin> read terminal-server  
TERMINAL-SERVER read
```


- 2 List the parameters in the Modem-Configuration subprofile. For example:

```
admin> list modem-configuration
v42/mnp = will-v42
max-baud-rate = 33600-max-baud
modem-transmit-level = -13-db-mdm-trn-level
cell-mode-first = no
cell-level = -18-db-cell-level
7-even = no
```

- 3 Modify the parameters as required.

For information about the parameters, see the *APX 8000/MAX TNT/DSL/TNT Reference*.

Specifying modem modulation for Series56 II and III modem cards

The Modem-Mod parameter in the Terminal-Server profile allows you to specify the modem modulation that Series56 II and III modems use. The possible settings are K56-Modulation, V34-Modulation, and V90-Modulation (the default).

To support the ITU-T standard V.8bis (Voice Call Ready), a 56Kbps modem in the APX 8000 and MAX TNT unit normally sends a tone at the beginning of modem training. This is commonly referred to as CRe and is a dual tone (1375Hz + 2002Hz) followed by a single tone at 400Hz with a combined duration of approximately 500ms. Although V.8bis is designed not to interfere with V.32bis modem negotiation, some V.32 and V.34 modems do not successfully complete modem training after reception of the V.8bis tone.

Note: If you configure the Series56 II and III modems to use V.34 modulation, they never exceed the speeds used by V.34 modems (33.6Kbps), and they do not send the V.8bis tone.

To configure modem modulation for calls coming in to Series56 II and III modem cards, proceed as in the following example:

```
admin> read terminal-server
TERMINAL-SERVER read

admin> set modem-configuration modem-mod = v34-modulation

admin> write
TERMINAL-SERVER write
```

Configuring an additional AT answer string for modem calls

The AT-Answer-String parameter in the Terminal-Server profile enables you to specify extra AT commands in the answer string of the system's modem configuration.

The answer string is the last of four strings that the APX 8000 or MAX TNT sends to the modem upon answering a call. Commands entered in this string might overwrite settings specified elsewhere. For example, if the Max-Baud-Rate parameter sets the maximum baud

rate and the AT-Answer-String parameter specifies a different baud rate, the answer string overwrites the configured maximum baud rate.

Following is the relevant parameter, which is shown with its default setting:

```
[in TERMINAL-SERVER:modem-configuration]
AT-answer-string = ""
```

The value of this parameter must be valid AT commands, up to 36 characters. Do not begin the string with AT. An AT is appended to the beginning of this string automatically before it is sent to the modem. Also, do not include an A (answer) or a D (dial) command anywhere in the string. An A command is appended automatically to the end of this string, and a D command in the answer string causes the call to fail.

Note: Be very careful when entering AT commands in this parameter. The system does not prevent you from entering incorrect strings.

The following example sets the AT-Answer-String parameter to S37=11, which causes the following string to be sent to the modem:

```
ATS37=11A
```

When the modem receives this string, it forces a V.32bis 14400 connection.

```
admin> read terminal-server
TERMINAL-SERVER read

admin> set modem AT-answer-string = S37=11

admin> write
TERMINAL-SERVER written
```

Series56 II and III Call-Route profiles

When you install a Series56 II or Series56 III slot card, the TAOS unit creates two call route profiles for each channel on the card. One for a digital data call and one for a modem voice call. For example:

```
admin >callroute -d
```

device	#	source	type	tg	sa	phone
1:14:01/0	0	0:00:00/0	voice-call-type	0	0	
1:14:01/0	1	0:00:00/0	digital-call-type	0	0	
1:14:02/0	0	0:00:00/0	voice-call-type	0	0	
1:14:02/0	1	0:00:00/0	digital-call-type	0	0	
1:14:03/0	0	0:00:00/0	voice-call-type	0	0	
1:14:03/0	1	0:00:00/0	digital-call-type	0	0	
1:14:04/0	0	0:00:00/0	voice-call-type	0	0	
1:14:04/0	1	0:00:00/0	digital-call-type	0	0	
1:14:05/0	0	0:00:00/0	voice-call-type	0	0	
1:14:05/0	1	0:00:00/0	digital-call-type	0	0	

Note that in Call-Route profiles, voice-call-type refers only to a modem call.

Preventing Series56 II and III cards from delaying Frame Relay connections

If the APX 8000 or MAX TNT has a Frame Relay datalink that uses a single nailed channel, you must install Series56 II or Series56 III slot cards in lower-numbered slots than the Hybrid Access (HDLC) slot cards, or dedicate the Series56 cards to modem processing by deleting the Digital Call-Type profiles. Otherwise, you are likely to experience delays in establishing Frame Relay connections. See Chapter 19, “Configuring Call Routing” for more information.

Hybrid Access card implementation

Each ISDN call, and each channel of a nailed session, requires an HDLC channel to process the HDLC-encapsulated data received from or destined to a WAN interface. Because the following cards require HDLC channels, you might need to install a Hybrid Access card in your unit:

- Eight-port E1 card
- Eight-port T1 card
- T3 card

The Hybrid Access card is supported on the DSLTNT. On an APX 8000 or MAX TNT unit, Series56 II and III cards also provide up to 48 HDLC channels per card.

Configuring MultiDSP Cards (MAX TNT, APX 8000)

Introduction to MultiDSP.	6-1
Supported MultiDSP services	6-3
Obtaining status information about a MultiDSP card	6-5
Configuring a MultiDSP card	6-6

Introduction to MultiDSP

The MultiDSP card is a highly versatile, digital signal processor (DSP) slot card for MAX TNT and APX 8000 units.

The following services are supported by the MultiDSP card:

- Data (digital, analog)
- V.110 rate adaption standard for ISDN
- Personal Handyphone System (PHS), supporting PHS Internet Access Forum Standards (PIAFS) 1.0, 2.0, 2.1
- Voice over IP (VoIP), including real-time fax functionality (MAX TNT only)

The MultiDSP support provided by a particular MultiDSP card or MAX TNT and APX 8000 unit depends on the following factors:

- The type of MultiDSP card(s) installed in the unit
- Software licenses (hash codes) currently downloaded on the unit shelf controller
- MultiDSP card use constraints (See “Card configuration constraints” on page 6-3 for details.)

Analog modem service is, by default, always enabled. Each additional MultiDSP service has a software license (hash code) that must be downloaded to the unit shelf controller for the particular service to be enabled on a MAX TNT or APX 8000 unit. The hash codes enable different types of calls to be serviced by the same DSP port on the card.

The MultiDSP services currently enabled (licensed) on MAX TNT and APX 8000 units can be viewed in the Base profile. For details, see the “Configuring a MultiDSP card” on page 6-6.

MAX TNT and APX 8000 units support two types of MultiDSP cards—a 48-port card and a 96-port card. Each MultiDSP card type supports slightly different services. Following are descriptions of each card.

48-port MultiDSP card

The 48-port MultiDSP card supports up to 48 ports of service.

Note: In MAX TNT and APX 8000 profiles and parameters, the 48-port MultiDSP card is identified as *madd* or *madd-card*.

For a MAX TNT or APX 8000 unit with a 48-port MultiDSP card, Lucent recommends that you limit the number of enabled MultiDSP services to two. Voice over IP (VoIP) is currently supported on the MAX TNT only.

When two services are supported by the card, one service must be data and the other can be V.110 PHS or VoIP. The following possible configurations are supported by the 48-port card:

- Data (analog and/or digital) service only
- V.110 service only
- PHS service only
- VoIP service only (MAX TNT only)
- Data and V.110 services
- Data and PHS services
- Data and VoIP services (MAX TNT only)

Downloaded software licenses (hash codes) determine which MultiDSP services are supported by a particular unit and 48-port MultiDSP card. For example, if a unit is licensed to run both data and VoIP, the ports on each installed 48-port MultiDSP card can handle data and/or VoIP calls.

96-port MultiDSP card

The 96-port MultiDSP card supports up to 96 ports of service.

Note: In MAX TNT and APX 8000 profiles and parameters, the 96-port MultiDSP card is identified as *madd2-card*.

A MAX TNT or APX 8000 unit with a 96-port MultiDSP card installed can have software licenses for up to two of the following MultiDSP services: data and V.110. The following possible configurations are supported by the 96-port card:

- Data (analog and/or digital) service only
- V.110 service only
- Data and V.110 services

Downloaded software licenses (hash codes) determine which MultiDSP services are supported by a particular unit and 96-port MultiDSP card. For example, if a unit is licensed to run both data and V.110, the ports on each installed 96-port MultiDSP card can handle data and/or V.110 calls.

Card configuration constraints

The following constraints affect the mixing of slot cards in MAX TNT and APX 8000 units.

Using 48-port and 96-port MultiDSP cards

You cannot mix 48-port and 96-port MultiDSP cards in the same MAX TNT or APX 8000 unit. However, you can use multiple 48-port or multiple 96-port MultiDSP in the same unit.

Using Series56 cards with MultiDSP cards

The single-slot Series56 II card and the single-slot Series56 III card can be used with a MultiDSP card in the same MAX TNT or APX 8000 unit. The dual-slot Series56 modem card cannot be used in a MAX TNT unit that has a MultiDSP card installed.

Supported MultiDSP services

The following sections describe the services (applications) supported by the MultiDSP card. See “Configuring a MultiDSP card” on page 6-6 for information on configuring these services in a unit.

Data

The MultiDSP card supports calls made through analog modems that comply with standards such as V.90, and digital calls made through the High-Level Data Link Control (HDLC) protocol. Digital calls can come from an ISDN Primary Rate Interface (PRI) line, a Signaling System 7 (SS7) network, or an E1 line with R2 signaling.

Although analog modem service is enabled by default, additional software licenses might be required to support digital calls that use particular signaling schemes or protocols. For example, a software license is required to support the R2 call setup signaling protocol. Also, software licenses are required to support the Ascend SS7 Gateway Control Protocol (ASGCP) and the IP Device Control (IPDC) protocol, which are the call setup intermachine trunk (IMT) protocols for SS7.

For information about configuring a unit for specific types of digital calls, refer to subsequent chapters in this guide and to the configuration guides listed under “Documentation set” on page xxi.

V.110

V.110 is a rate adaption standard that allows telephones using the digital cellular Global System for Mobile Communication (GSM) to connect to an ISDN network.

The V.110 service is supported by both the 48-port and 96-port MultiDSP card and requires a V.110 software license. V.110 support also requires a software license for the associated digital signaling scheme, which can be PRI, R2, ASGCP IMT/SS7, or IPDC IMT/SS7.

V.110 features supported by the MultiDSP card include the following:

- Asynchronous, answer-mode only (answer, but no call out), with 1 start bit, 8 data bits, and 1 stop bit.
- Rate-adaptive mode. Supported rates are 2400bps, 4800bps, 9600bps (default), 19200bps and 38400bps.

PHS

The Personal Handyphone System (PHS) provides mobile telephone access to users located in Japan and other Asian countries. PHS provides data communication services at bandwidths up to 64Kbps and offers voice communication services.

The MultiDSP card supports the following data PIAFS standards:

- PIAFS 1.0: Fixed data rate of 32Kbps.
- PIAFS 2.0: Fixed data rate of either 32Kbps or 64Kbps for the duration of a call.
- PIAFS 2.1: Data rate that can dynamically switch between 32Kbps and 64Kbps during a call, depending on the available wireless bandwidth.

A PHS software license is required to support PIAFS 1.0 and 2.0 functionalities. The license supports fixed data rates of 32Kbps or 64Kbps. The data rate used by the unit is determined by the rate from the PRI line. PHS service is currently available only with Japan PRI signaling.

To support PIAFS 2.1 functionality, two software licenses are required—the initial PHS software license for PIAFS 1.0 and PIAFS 2.0, and a separate PHS PIAFS 2.1 software license. The two licenses are also available bundled into one license package.

Voice over IP (VoIP)

VoIP is a service that offers voice telephony across IP network infrastructures. The MultiDSP VoIP implementation relies on the MultiVoice Gateway to connect calls to public and private packet networks. The MultiDSP card's VoIP implementation supports the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) standard for H.323 signaling and messaging.

VoIP features supported by the MultiDSP card include the following:

- ITU-T H.323 signaling and messaging.
- Voice compression and packetization.
- Connection of each port to a single DS0 (voice calls).
- Cut-through of progress tone signals from the distant Public Switched Telephone Network (PSTN).
- Encoding schemes G.711 A-law, G.711 μ -law, G.723.1, G.728, G.729, and RT-24 (a codec (coder/decoder) developed by Lucent).
- Silence suppression and detection for G.729, configured through a MAX TNT unit's VoIP profile. Silence suppression is automatically enabled for G.723.1.
- Real-time fax (T.38 fax).

A VoIP software license is required for MultiDSP support of the VoIP service. An additional software license is required for support of real-time fax functionality.

VoIP functionalities, including real-time fax, are configured through the MAX TNT VoIP profile.

For details about VoIP and MultiVoice configuration, refer to the *MultiVoice for MAX TNT Configuration Guide*. In addition, see “Configuring a MultiDSP card” on page 6-6.

Obtaining status information about a MultiDSP card

Information can be displayed about all installed cards or only an installed MultiDSP card.

Displaying information about all installed cards

The `show` command displays the following information about currently installed slot cards, including the MultiDSP card:

- Location of all installed cards by shelf number, slot number, and item or port number, including the MultiDSP card
- Status of each card (for example, if the card is Up or Down)
- Type of cards currently installed

Note: The 48-port MultiDSP card is displayed with the name *madd* or *madd-card*. The 96-port MultiDSP card is displayed with the name *madd2-card*.

Use the `show` command to confirm that the MultiDSP card is listed as one of the installed cards and is shown installed in the correct slot. For example, in a MAX TNT unit the command displays the following information if the unit has only one shelf (shelf 1) and a 96-port MultiDSP card is installed in slot 3:

```
admin> show
Shelf 1 ( standalone ):
{ shelf-1 slot-3 0 }      UP      madd2-card
```

Displaying information about an installed MultiDSP card

Use the `show` command with the MultiDSP card shelf number and slot number to display the following information about a particular installed MultiDSP card:

- The card's ports by shelf number, slot number, and port number
- Status of each port (for example, whether the port is Up or Down)
- The port service type (for example, modem)

Enter the `show` command, along with the shelf number and port number, to confirm that all MultiDSP card ports are shown in the listing.

Note: For the 96-port MultiDSP card, the `show shelf slot` command displays 96 modems (ports). For the 48-port card, only the odd-numbered modems (1, 3, ..., 95) are active. The `modem -a` command verifies that only 48 modems are available on the 48-port card.

For example, enter the following command to display information about a 96-port MultiDSP card (identified as *madd2-card*) installed in shelf 1, slot 10 of a MAX TNT unit:

```
admin> show 1 10

{ shelf-1 slot-10 0 }      UP      madd2-card:
{ shelf-1 slot-10 1 }      UP      madd-modem-1
{ shelf-1 slot-10 2 }      UP      madd-modem-2
{ shelf-1 slot-10 3 }      UP      madd-modem-3
...
{ shelf-1 slot-10 96 }     UP      madd-modem-96
```

Verifying that installed software and software versions are correct

The `dircode` command displays descriptions and version numbers of all software currently installed on the unit's flash memory card.

Use the `dircode` command to verify that the software version numbers for the system (shelf controller) and MultiDSP card (shown as *madd-card*) are correct.

The following example shows the installed software versions for a MAX TNT system (shelf controller) and 48-port MultiDSP card:

```
admin> dircode

Flash card code directory:
Card 1, directory size 16

shelf-controller  1838073 Thu Jan 6 19:15:52 2000 Version 8.0.0
madd-card         1336282 Thu Jan 6 19:16:00 2000 Version 8.0.0
```

Configuring a MultiDSP card

When a unit detects the presence of a card in one of its slots, the unit creates default profiles appropriate for that type of card. Some default profiles might need to be reconfigured when a new card is installed in a slot.

Minimal configuration is required to set up MultiDSP services on a unit. Licensed MultiDSP services are automatically enabled. Depending on the type of MultiDSP card installed, 48 or 96 default call routes are created for each enabled service when the card is installed.

Configuration might be required for implementing VoIP MultiVoice functionality. VoIP MultiVoice configuration is described in the *MultiVoice for MAX TNT Configuration Guide*.

MultiDSP configuration includes the following tasks:

- Verifying that the correct software licenses are loaded on the MAX TNT or APX 8000 unit and that the desired MultiDSP services are enabled. The Base profile provides this information.
- Confirming that the call routes are correct for the desired MultiDSP services. The Call-Route profile controls call routing in the unit. The unit automatically generates call routes for modem calls and licensed services when a MultiDSP card is installed or when the `slot -r` and `slot -u` commands are issued.

- Verifying that all other configurations associated with the MultiDSP services have been performed. For example, for VoIP MultiDSP support, all required configurations for VoIP, real-time fax (optional), and other desired MultiVoice features must also be performed.
- Adding support for an additional MultiDSP service, if necessary.

Verifying that MultiDSP services are enabled

The Base profile is a read-only, system-wide profile that displays enabled features, network interfaces, and system information. Included in the Base profile are the MultiDSP services. The Base profile indicates which MultiDSP services are currently enabled (licensed). Because the MultiDSP modem function does not require a software license, no modem-related parameter appears in the Base profile.

Display the Base Profile to verify that desired services are licensed on the unit. Use the `get base` command to view the profile.

The following Base profile parameters relevant to the MultiDSP card. Verify that the appropriate services are supported for your MultiDSP applications.

Base profile parameter	Value if supported
Data-Call-Enabled	Yes if the unit supports data calls over ISDN (digital) lines. The parameter is also used by Series56 and Hybrid Access (HDLC) cards.
R2-Signaling-Enabled	Yes for R2 signaling support.
SS7-ASG	Yes for ASGCP SS7 IMT signaling support.
XCOM-SS7	Yes for IPDC SS7 IMT signaling support.
V110-Enabled	Enabled if V.110 software is licensed on the unit.
PHS-Support	Yes if Personal Handyphone System PIAFS 1.0, PIAFS 2.0 support is licensed on the unit. For PIAFS 2.1 support, the PHS-2-1-Support parameter must also be enabled by a software license.
PHS-2-1-Support	Yes if PIAFS 2.1 support is licensed on the unit. The PHS-Support parameter must also be enabled by a software license. (The two PHS licenses are also available bundled into one package.)
VoIP-Enabled	Yes if VoIP is enabled by a software license.
RTFax-Enabled	Yes if real-time fax (T.38) is licensed. For real-time fax support, the VoIP-Enabled parameter must also be enabled by a separate software license.

The following example shows relevant Base profile parameters and values for a unit installed with a 48-port MultiDSP card that supports modem calls, digital HDLC calls, and PHS PIAFS 1.0 and PIAFS 2.0 calls (but not PIAFS 2.1 calls):

```
admin> get base
...
data-call-enabled = yes
...
phs-2-1-support = no
```

```
...
phs-support = yes
...
voip-enabled = no
...
v110-enabled = disabled
...
rtfax-enabled = no
...
```

Verifying call routes for MultiDSP services

You can verify call routes for enabled MultiDSP services by viewing the Call-Route profiles and the call route entries. Procedures for viewing the Call-Route profiles and entries are described in this section.

Viewing the Call-Route profile and its Call-Route-Type parameter

When a 48-port or 96-port MultiDSP card is initially detected by the unit, the unit creates Call-Route profiles for each MultiDSP service—modem, digital, PHS, VoIP or V.110.

Use the `dir call-r` command to display the Call Route profiles. The following example displays Call Route profiles for a 48-port card (indicated by *madd*) installed in slot 5 of a MAX TNT unit:

```
admin> show
Shelf 1 ( standalone ):
{ shelf-1 slot-5 0 }    UP    madd-card

admin> dir call-r

30 12/07/1999 10:22:12 {{{shelf-1 any-slot 0} 0} 0}
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 0}
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 1}
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 2}
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 3}
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 4}
```

The first entry is the system default. The other profiles are for each of the MultiDSP services. You can edit the profile list to include only currently enabled MultiDSP services.

Each profile contains an index that uses the following format:

```
{{{shelf slot port} logical-item} entry}
```

The system index is the following:

```
{{{shelf-1 any-slot 0} 0} 0}.
```

Each MultiDSP service has a unique *entry* field number. The following table displays the *entry* number associated with each MultiDSP service type:

Call-Route profile <i>entry</i> field number	Associated MultiDSP service type
0	Analog modem

Call-Route profile entry field number	Associated MultiDSP service type
1	Digital
2	PHS
3	VoIP
4	V.110

The following example shows a Call-Route profile for the VoIP service (*entry* is 3) on a 48-port MultiDSP card installed in slot 5:

```
33 12/10/1999 18:25:32 {{{shelf-1 slot-5 0} 0} 3}
```

About the Call-Route-Type parameter

Each Call-Route profile contains the Call-Route-Type parameters specific to that profile. The Call-Route parameters might require configuration.

One Call-Route parameter, Call-Route-Type, specifies the type of call that the MAX TNT can route to a host device. The following Call-Route-Type values apply to MultiDSP services.

Call-Route-Type values for MultiDSP services	Description
Voice-Call-Type	Call type for analog mode calls. The unit can route voice bearer calls, excluding 3.1KHz audio calls, to a host device.
Digital-Call-Type	Call type for digital calls. The unit can route digital calls, including 3.1KHz audio bearer channels, to a host device.
PHS-Call-Type	Call type for PHS calls.
VOIP-Call-Type	Call type for VoIP calls. VoIP calls can be routed to a host device that accepts VoIP calls.
V110-Call-Type	Call type for V.110 calls. Digital calls recognized as containing V.110 rate-adapted bearer channels can be routed to a host device.

Use the `read call-route` and `list` commands to view a specific profile's parameters. To identify the specific profile, you must include the profile's index. The following example shows Call Route parameters for the profile that covers the V.110 service (*entry* field is 4), for a unit with a MultiDSP card in slot 5:

```
admin> read call-route { { { shelf-1 slot-5 0 } 0 } 4 }
CALL-ROUTE/{ { { shelf-1 slot-5 0 } 0 } 4 }

admin> list
[in CALL-ROUTE/{ { { shelf-1 slot-5 } 0 4 }}
index* = { { { shelf-1 slot-5 0 } 0 } 4 }
trunk-group = 0
phone-number = ""
preferred-source = { { any-shelf any-slot 0 } 0}
call-route-type = v110-call-type
```

Viewing call-routing database entries

Unlike the Call Route profiles, entries in a call-routing database are created for the analog modem service, digital service, and for each licensed MultiDSP service.

When a card comes up, the MAX TNT or APX 8000 unit creates a call-routing database. The number of database entries created per service depend on the following:

- Type (48-port or 96-port) of MultiDSP card being used.
- Enabled MultiDSP services. (Analog modem service is always enabled.)

A call route entry is created for each available port and each enabled service. To view the entries in the call-routing database, use the `callroute -a` command.

Note: For the 48-port MultiDSP card, only the 48 odd-numbered ports are available (port 1, 3, 5, ..., 95).

For example, if a 48-port MultiDSP card is installed in slot 5 of the MAX TNT and the supported MultiDSP services are data (analog modem and digital) and PHS, 48 call route entries are created for analog modem service, 48 call route entries are created for digital service, and 48 call route entries are created for PHS service:

```
admin> callroute -a
1:05:01/0    0 0:00:00/0    voice-call-type    0 0
1:05:03/0    0 0:00:00/0    voice-call-type    0 0
...
1:05:95/0    0 0:00:00/0    voice-call-type    0 0
1:05:01/0    1 0:00:00/0    digital-call-type   0 0
1:05:03/0    1 0:00:00/0    digital-call-type   0 0
...
1:05:95/0    1 0:00:00/0    digital-call-type   0 0
1:05:01/0    2 0:00:00/0    phs-call-type       0 0
1:05:03/0    2 0:00:00/0    phs-call-type       0 0
...
1:05:95/0    2 0:00:00/0    phs-call-type       0 0
```

Verifying that configurations are correct for related services

Each MultiDSP service might require additional configuration for setting up that service on the unit. For additional information and procedures, see the configuration guides listed under “Documentation set” on page xxi and the latest release note.

Adding an additional MultiDSP service

To enable an additional MultiDSP service, you must perform the following steps:

- 1 The unit must have the proper software license for the desired service. For further information about adding MultiDSP software licenses, contact your Lucent Sales Representative.
- 2 Once the unit is licensed for the new service, delete the existing profiles for MultiDSP cards installed in the unit, and bring up the current profiles (that now include the new license).

For example, if you want to delete the profiles and then bring up the current profiles for a card installed in slot 5, enter the following commands:

```
admin> slot -r 1 5  
admin> slot -u
```

When the MultiDSP cards are brought up, the unit creates new profiles and call route entries for each service, including the new service.

Configuring T1 Cards

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Introduction to T1

A T1 line consists of 24 channels. Each channel can transmit and receive data or digitized voice. The line uses framing and signaling to achieve synchronous and reliable transmission. The most common configurations for T1 lines are ISDN Primary Rate Interface (PRI) and nailed (leased) or unchannelized T1, including fractional T1. (For information about provisioning your T1 line for use with the TAOS unit, see Appendix A, “Provisioning the Switch.”)

ISDN PRI

In North America and Japan, a T1/PRI line typically supports 23 B channels and one D channel. But if non-facility associated signaling (NFAS) is in use, more than one ISDN PRI line on a single T1 card can share a single D channel. PRI configurations are used to receive multiple, simultaneous ISDN calls from analog-modem and digital-services dial-in traffic. Another common use of T1/PRI is to connect a Private Branch Exchange (PBX) to a central office switch.

Nailed or unchannelized T1

Unchannelized T1 lines can be used for nailed connections such as to a Frame Relay network. In such cases the configuration is static, and the TAOS unit treats the T1 line as if it were a single connection at a fixed speed, without individual channels.

Typically, when you pay your telephone company for a leased (nailed) line, you pay more for higher bandwidth. Anything in the range of 0bps to 1.544Mbps can be delivered on a T1 line, and provisioned at some 64Kbps fraction of the full T1 bandwidth.

Channelized line-side vs. trunk-side T1

Calls entering the telephone network from the TAOS unit must enter the central office (CO) through an ISDN PRI line. However, calls coming in on a channelized T1 line can enter either on the line side or trunk side. For best results, ensure that the channelized T1 calls enter the switch on the trunk side.

T1 lines that terminate on the line side of the switch undergo an additional analog-to-digital conversion that reduces the data transfer rate. Some service providers and carriers have agreements to ensure that a T1 always enters the trunk side of the CO switch, but in most cases, no such agreement exists. The only way to guarantee a digital connection is to make sure that calls from the TAOS unit enter the CO on the trunk side of the switch over an ISDN PRI or a trunk-side T1 line.

Overview of T1 configuration

Table 7-1 lists the sections describing common tasks you might have to perform to configure a T1 line. The table includes a brief description of each task and lists the parameters you will use.

For information about administering the T1 card, including such tasks as specifying a facilities data link (FDL) and displaying the status of the lines, see the *APX 8000/MAX TNT/DSLNT Administration Guide*.

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Table 7-1. T1 line configuration tasks

Section	Description of task	Associated parameters
“Making a profile the working profile” on page 7-6	Before you can edit a profile, you must make it the working profile.	N/A
“Assigning names to T1 line profiles” on page 7-7	Assign a name to the T1 profile.	Name
“Enabling a line” on page 7-8	Make a line available for use.	Enabled
“Specifying the framing and encoding” on page 7-8	Each T1 line requires framing and encoding. Framing specifies the format for the sequence of bits sent on the line. Encoding affects the way data is represented by the digital signals on the line.	Frame-Type Encoding
“Configuring ISDN PRI signaling” on page 7-8	You must specify the type of network switch providing ISDN service on a T1 PRI line.	Switch-Type
“Configuring ISDN network-side emulation” on page 7-9	ISDN emulation enables you to build, send, receive, and process ISDN data.	ISDN-Emulation-Side
“Configuring overlap receiving on PRI lines” on page 7-9	T1 or E1 PRI lines with overlap receiving enable the TAOS unit to gather the complete called number from the network switch via a series of Information messages, enabling the use of features such as called-number authentication.	Signaling-Mode Overlap-Receiving PRI-Prefix-Number Trailing-Digits T302-Timer

Table 7-1. T1 line configuration tasks (continued)

Section	Description of task	Associated parameters
“Configuring inband robbed-bit signaling” on page 7-11	If the lines use inband signaling, change the signaling mode to robbed bit and specify the type of robbed bit signaling to use. You can also specify that the TAOS unit process the numbers dialed for use with Dialed Number Identification Service (DNIS) and Calling Number Identification (CLID) authentication.	Signaling-Mode Robbed-Bit-Mode Collect-Incoming-Digits DSP-DTMF-Input-Sample-Count
“Configuring NFAS” on page 7-13	Specify non-facility associated signaling (NFAS) if you want two or more PRI lines to share a D channel.	Switch-Type NFAS-ID NFAS-Group-ID
“Configuring T1 R1 and R1-Modified (Taiwan) with ANI and called-number processing” on page 7-15	R1 is a multifrequency inband signaling system that uses a set of register signals known as MFR1 tones as addressing signals. R1 signaling can optionally be used with Automatic Number Identification (ANI), which is similar to Caller ID (CLID).	Signaling-Mode R1-Use-ANIR R1-First-Digit-Timer R1-ANIR-Delay R1-ANIR-Timer R1-Modified
“Configuring clocking” on page 7-17	Set Clock-Source to specify whether the T1 line can be used as the master clock source for synchronous connections. Also specify the priority of the T1 lines to be used for clocking.	Clock-Source Clock-Priority
“Configuring the front-end transceiver” on page 7-17	Set the front-end type of the T1 transceiver to CSU (channel service unit) or DSX (digital signal cross connect), depending on the type of device the TAOS unit connects to.	Front-End-Type DSX-Line-Length CSU-Build-Out
“Configuring channel usage” on page 7-18	Specify how each of the 24 channels of a T1 line is to be used.	Channel-Usage
“Assigning telephone numbers to switched channels” on page 7-19	Typically, you specify only the rightmost digits necessary to distinguish one number from another. These are called add-on numbers.	Phone-Number
“Configuring trunk groups” on page 7-20	A trunk group is a group of channels that has been assigned a number.	Trunk-Group

Table 7-1. T1 line configuration tasks (continued)

Section	Description of task	Associated parameters
“Configuring nailed channels” on page 7-21	You must assign a nailed channel to a group to make it available for use. The group number can be referred to in a Connection or Frame-Relay profile to specify a permanent leased connection using that group of nailed channels.	Nailed-Group
“Configuring a back-to-back T1 connection” on page 7-21	For diagnostic purposes you might sometimes want to configure a back- to-back connectionT1 connection between ports on two TAOS unit units.	Signaling-Mode set to Inband (the default) Robbed-Bit-Mode set to Wink-Start (the default) Clock-Source set to Eligible (the default)
“Specifying analog encoding for TAOS unit codecs” on page 7-22	Codecs connected to T1 use a different encoding standard for digitized analog data than do codecs connected to E1. The default for T1 is U-Law, the default for E1 is A-Law.	Analog-Encoding
“Configuring specialized options” on page 7-22	Typically, the D channel of a PRI line uses normal data. However, for some connections, you might need to invert the data to avoid transmitting a pattern that the connection cannot handle Most installations use the default for the Idle-Mode setting, which determines what pattern the D channel looks for to specify the idle indicator.	Data-Sense Idle-Mode
Chapter 19, “Configuring Call Routing”	The TAOS unit uses call routing to determine where to route incoming and outgoing calls. The preferred way to set up call-routing is to put all call-routing information in one place: a Call-Route profile. If you do not use Call-Route profiles, specify the physical address of a device to which calls received on this channel are routed.	Default-Call-Type Call-by-Call-Service Shelf Slot Item

Making a profile the working profile

When the TAOS unit detects that a T1 card has been installed, it creates a default T1 profile for each of the eight lines on the card.

In the following display example, the Dir command shows eight default T1 profiles created for a card installed in slot 2:

```
admin> dir t1
305 12/11/1996 15:58:20 { shelf-1 slot-2 2 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 4 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 5 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 6 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 7 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 8 }
320 12/20/1996 20:55:31 { shelf-1 slot-2 3 }
317 01/08/1997 09:58:55 { shelf-1 slot-2 1 }
```

By default, the line is not enabled, which means that it is not available for use. Its default signaling method is inband, typically used for channelized connections.

To configure a T1 profile, first make it the working profile by reading it into the edit buffer. For example:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read
```

Once you have read in a profile, it remains the working profile until you read in another profile. You can use the Set command to change one or more of the profile's parameters.

To save your configuration changes, use the Write command. For example:

```
admin> write
T1/{ shelf-1 slot-2 1 } written
```

To list the parameters in a T1 profile, use the List command, as in the following example:

```
admin> list
[in T1/{ shelf-1 slot-6 4 }]
name = " "
physical-address* = { shelf-1 slot-6 4 }
line-interface = { no d4 ami eligible low-priority inband +
```

The following example shows the parameters in a T1 profile:

```
[in T1/{ shelf-1 slot-6 4 }:line-interface]
enabled = no
frame-type = d4
encoding = ami
clock-source = eligible
clock-priority = low-priority
signaling-mode = inband
robbed-bit-mode = wink-start
default-call-type = digital
switch-type = att-pri
```

```
nfas-group-id = 0
nfas-id = 0
incoming-call-handling = internal-processing
call-by-call = 0
data-sense = normal
idle-mode = flag-idle
FDL = none
front-end-type = dsx
DSX-line-length = 1-133
CSU-build-out = 0-db
overlap-receiving = no
pri-prefix-number = ""
trailing-digits = 2
t302-timer = 10000
channel-config = [ { unused-channel 9 "" { any-shelf
any-slot +
maintenance-state = no
input-sample-count = one-sample
sendDisc-val = 0
hunt-grp-phone-number-1 = ""
hunt-grp-phone-number-2 = ""
hunt-grp-phone-number-3 = ""
collect-incoming-digits = no
r1-use-anir = no
r1-first-digit-timer = 340
r1-anir-delay = 350
r1-anir-timer = 200
r1-modified = no
```

Assigning names to T1 line profiles

In a T1 profile, the Name parameter enables you to assign the profile a name. The name can include up to 16 characters. It is displayed after the line's physical address in the Dir command output. For example:

```
admin> read t1 {1 12 0}
admin> set name = T1 Trunk
admin> write
T1/{ shelf-1 slot-12 0 } written
admin> dir T1
17  04/17/1997 19:00:02  { shelf-1 slot-12 0 }  "T1 Trunk"
```

For T1 lines, the Line Status window displays the first eight characters of the name if one has been assigned. For example:

```
T1 Trunk 1/12/0  LA la  la  la  la  la  la  la
```

If the name is longer than eight characters, the last character displayed is a plus sign (+).

Enabling a line

By default each T1 line is disabled. To enable the T1 line, read its profile to make it the working profile, then set the Line Interface subprofile's Enabled parameter to Yes, as in the following example:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read
admin> set line enabled = yes
admin> write
T1/{ shelf-1 slot-2 1 } written
```

Specifying the framing and encoding

You must specify the framing and the encoding for each T1 line. If you are using ISDN, you must specify the extended superframe (ESF) format, which consists of 24 consecutive frames, separated by framing bits. If the line is not configured for ISDN signaling, use D4 framing (also known as the superframe format), which is the default.

The T1 Encoding value sets the layer-1 line encoding used for the physical links, which affects the way in which data is represented by the digital signals on the line. The default, alternate mark inversion (AMI) encoding, is often used, although bipolar with 8-zero substitution (B8ZS) encoding might be required if the line is configured for ISDN signaling. If set to None, encoding is similar to AMI, but without density enforcement.

Your T1 service provider must provide the correct framing and encoding values for your lines.

To specify the framing and encoding, set the Frame-Type and Encoding parameters:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read
admin> set line frame-type = [esf|d4]
admin> set encoding = [ami|b8zs|none]
admin> write
T1/{ shelf-1 slot-2 1 } written
```

Configuring ISDN PRI signaling

When you set the signaling mode to ISDN, you must set channel 24 as the D channel. Note that ISDN signaling often requires ESF framing and B8ZS encoding.

For ISDN signaling you must also specify the type of switch providing T1/PRI service to your TAOS unit. Obtain the information from your ISDN carrier. (For example, if your carrier is AT&T, the switch type is ATT-PRI.)

Configure ISDN PRI service as follows:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read
admin> set line frame-type = esf
admin> set line encoding = b8zs
admin> set line signaling-mode = isdn
```



```
admin> set line switch-type = switchtype
admin> set line channel 24 channel-usage=d-channel
admin> write
T1/{ shelf-1 slot-2 1} written
```

To see a complete list of switch types supported on the TAOS unit, refer to the TAOS unit online help or the *APX 8000/MAX TNT/DSLNT Reference*.

Configuring ISDN network-side emulation

You can configure PRI lines to use either network-side or user-side ISDN emulation. Previously, PRI lines on the TAOS unit supported only user-side emulation. Following is the relevant parameter, shown with its default setting:

```
[in T1/{ any-shelf any-slot0 }:line-interface]
isdn-emulation-side = te
```

ISDN is a nonsymmetrical protocol used by telephone carriers to provide digital services to end users. There are no ISDN links between telephone carrier Central Offices (COs). ISDN links exist only between the CO and the customer. Therefore, an ISDN link can be viewed as having two sides—the network side, or network terminating (NT) equipment, and the user side, or terminal equipment (TE). The user side can connect only to the network side, and vice versa. Both the network side and the user side perform the same functions, but the format of the messages is different. For example, the network side must always set a bit and the user side must always clear it. These differences allow either side to determine whether the other end is the right one.

ISDN emulation enables you to build, send, receive, and process ISDN data. ISDN monitoring, on the other hand, allows you only to decode the ISDN data.

Configuring overlap receiving on PRI lines

Overlap receiving affects the procedure of establishing an incoming call received on a T1 or E1 PRI line in the TAOS unit. With overlap receiving, the TAOS unit can gather the complete called number from the network switch via a series of Information messages, enabling the use of features such as called-number authentication.

The Q.931 specification states that either en-bloc receiving or overlap receiving can be used to handle an incoming call. With en-bloc receiving, the Setup message received from the network switch must contain all information required to process the call. With overlap receiving, the Setup message can contain incomplete called number information, with the remainder of the call information (if any) sent in one or more additional Information messages after the network switch receives a Setup Acknowledge message from the called unit.

Following are the relevant parameters, which are shown with sample settings for T1 and E1 lines:

```
[in T1/{ shelf-1 slot-5 1 }:line-interface]
signaling-mode = isdn-nfas
overlap-receiving = yes
pri-prefix-number = 3069
```

Configuring T1 Cards

Configuring overlap receiving on PRI lines

```
trailing-digits = 2
t302-timer = 10000

[in E1/{ shelf-1 slot-12 1 }:line-interface]
signaling-mode = isdn
overlap-receiving = yes
pri-prefix-number = 3069
trailing-digits = 2
t302-timer = 10000
```

To configure overlap receiving, you need to set some or all of the following parameters:

Parameter	Specifies
Signaling-Mode	Type of signaling on the T1 or E1 line. It must specify ISDN (or ISDN-NFAS, for T1) to use overlap receiving. If it is set to any other value, Overlap-Receiving does not apply.
Overlap-Receiving	Enables/disables overlap receiving for incoming calls on the PRI line. If set to No (the default), the PRI-Prefix-Number, Trailing-Digits, and T302-Time parameters do not apply for overlap receiving.
PRI-Prefix-Number	<p>Portion of the line's telephone number to be used when matching the called-party number in the Setup message from the network switch. The reason for specifying this number is to enable the TAOS unit to quickly determine when the called-party number is complete when overlap receiving is in use. The unit uses this number and the specified number of trailing digits to recognize that the called-party number is complete, even if the caller did not include a Sending Complete code (for example, by dialing the pound sign).</p> <p>Typically, the PRI prefix is an ISDN-subscriber number, that might include an area code or an area and country code combination (which must be separated from the ISDN-subscriber number by a hyphen). With this additional information, the TAOS unit looks for just the first match of PRI-Prefix-Number against the called-party number in the Setup message (first with area code, and if that fails, then without area code).</p> <p>The default null value disables the T302-Timer optimization.</p>
Trailing-Digits	<p>Number of digits required to follow the prefix number for the TAOS unit to consider the called number complete. Callers can indicate Sending Complete by a method such as dialing the pound sign (#). If a caller does not indicate Sending Complete and the TAOS unit cannot determine whether the called number is complete, the TAOS unit waits until the T302 timer expires even if the caller has dialed all the required digits.</p> <p>The Trailing-Digits setting enables the TAOS unit to reset the timer when the specified number of digits has been received. Trailing-Digits can specify a value from 1 to 6. The default value is 2.</p>

Parameter	Specifies
T302-Timer	<p>Number of milliseconds that the system waits for additional called number information for an incoming call. The valid range is from 100 to 30000 (.10 second and 30 seconds).</p> <p>The default is 10000 (0.10 seconds).</p> <p>The TAOS unit begins collecting the trailing digit information, and for each call Setup message from the switch that does <i>not</i> include "Sending Complete Information Element," it starts the T302 timer (the Setup Ack timer).</p> <p>The TAOS unit stops the timer when it receives a message that includes "Sending Complete Information Element." The TAOS unit assumes there are no more trailing digit digits to collect when the T302 timer stops or expires.</p>

The following example enables overlap receiving on an E1 PRI line:

```
admin> read e1 {1 16 7}
E1/{ shelf-1 slot-16 7 } read
admin> set signaling-mode = isdn
admin> set overlap-receiving = yes
admin> set pri-prefix-number = 049-228-555
admin> set trailing-digits = 4
admin> set t302-timer = 5000
```

With this configuration, if a caller dials 049-228-555-1212, the TAOS unit matches the prefix, finds four trailing digits, and immediately begins processing the call. It might use called-number authentication (if applicable) before establishing a session. Similarly, if a local caller dials 555-1212, the TAOS unit fails the first match, tries without the country code and fails again, tries without the area code, and succeeds. It then finds four trailing digits and begins processing the call.

Configuring inband robbed-bit signaling

When the line is configured for inband signaling, the TAOS unit does not receive bearer-capability information from the carrier. Therefore, it cannot determine when a call is voice-service or digital-service. For call-routing purposes, all calls in inband lines are treated as digital calls. You can change this default by setting the Default-Call-Type parameter.

Trunk-side T1 lines must use wink-start call control, which is the default. It enables the switch to seize the trunk by going off hook after receiving a 200ms wink.

Line-side T1 lines must use loop-start call control. Regardless of the type of call control mechanism you choose, the switch must not forward dialed digits to the TAOS unit. Doing so disrupts the handshaking process during multichannel calls. Lucent recommends that channelized T1 lines be trunk side rather than line side.

On lines configured for inband signaling, you must specify that the TAOS unit process the calling and called dual-tone multifrequency (DTMF) digits if you want to use Dialed Number Identification Service (DNIS) and Calling Number Identification (CLID) authentication or

accounting. (On lines configured for PRI signaling, this information is presented as part of the call setup message and does not require special configuration on the TAOS unit.)

To configure the TAOS unit to process the DTMF digits in a call, use the Collect-Incoming-Digits and DSP-DTMF-Input-Sample-Count parameters in a T1 profile.

The Collect-Incoming-Digits parameter enables the TAOS unit to process the DTMF digits in a call. The DSP-DTMF-Input-Sample-Count parameter specifies the number (one or two) of Goertzel input samples that the TAOS unit computes to decode a DTMF digit. A setting of Two-Samples creates a more accurate result.

To configure a T1 line for inband (robbed-bit) signaling, proceed as in the following example:

- 1 Read in the T1 profile:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read
```

- 2 List the Line-Interface subprofile:

```
admin> list line
enabled=no
frame-type=d4
encoding=ami
clock-source=eligible
clock-priority=middle-priority
signaling-mode=inband
robbed-bit-mode=wink-start
default-call-type = digital
collect-incoming-digits = no
dsp-dtmf=input-sample-count=one-sample
..
..
```

- 3 Enable the line:

```
admin> set enabled = yes
```

- 4 Specify inband signaling:

```
admin> set signaling-mode = inband
```

- 5 Specify the Robbed-Bit-Mode:

```
admin> set robbed-bit-mode = wink-start
```

- 6 Specify call type:

```
admin> set default call type = voice
```

- 7 If you are using DNIS or CLID authentication, set the TAOS unit to process the DTMF digits and specify the sample size used to decode the digits:

```
admin> set collect-incoming-digits = yes
admin> set dsp-dtmf-input-sample-count = one-sample
```

- 8 Write the profile to save the changes:

```
admin> write
T1/{ shelf-1 slot-2 1 } written
```

Configuring NFAS

A group of T1 lines configured for NFAS signaling shares a D channel. One line in the group is configured with a primary D channel, and another line is configured with a secondary D channel. The secondary D channel is used only if the primary line fails or receives a signal commanding a change to the other D channel. All lines within an NFAS group must reside on the same slot card. Your service provider must supply you with the NFAS ID numbers for your line.

The TAOS unit supports multiple NFAS groups on a single card. An NFAS group contains a minimum of two PRIs. A T1 card supports up to four NFAS groups, and a T3 card supports up to 14 NFAS groups. To configure an NFAS group, you must set the NFAS-group-ID parameter. Lines with the same NFAS-group-ID value are in the same NFAS group.

Configuring a single NFAS group

To configure two T1 lines for NFAS, proceed as in the following example, in which the administrator configures ports 3 and 4 of the card in slot 2 of shelf 1:

```
admin> read t1 {1 2 3}
T1/{ shelf-1 slot-2 3 } read

admin> set line enabled = yes
admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 0
admin> set channel 24 channel = nfas-primary
admin> write
T1/{ shelf-1 slot-2 3 } written

admin> read t1 {1 2 4}
T1/{ shelf-1 slot-2 4 } read

admin> set line enabled = yes
admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 1
admin> set line channel 24 channel = nfas-secondary-d
admin> write
T1/{ shelf-1 slot-2 4 } written
```

Configuring multiple NFAS groups

To configure multiple NFAS groups, you must first obtain an NFAS ID for each DS1 from your service provider and an NFAS group ID for each group of PRI lines that shares a D channel. Within an NFAS group, all PRIs share the same NFAS-group-ID value and have unique NFAS-ID values.

Telcos often use NFAS-ID=0 for the PRI with the primary D-Channel, and NFAS-ID=1 for the PRI with the secondary D channel. You must set both the NFAS-group-ID parameter and the NFAS-ID parameter for each DS1.

In the following example, an administrator configures two NFAS groups on a T1 card. Each group contains four DS1s. The example uses the NFAS group IDs 1 and 2, but the actual values you use depend on how your lines are provisioned.

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-2 1 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 0
admin> set line nfas-group-id = 1
admin> set channel 24 channel = nfas-primary
admin> write
T1/{ shelf-1 slot-2 1 } written

admin> read t1 {1 2 2}
T1/{ shelf-1 slot-2 2 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 1
admin> set line nfas-group-id = 1
admin> set line channel 24 channel = nfas-secondary
admin> write
T1/{ shelf-1 slot-2 2 } written

admin> read t1 {1 2 3}
T1/{ shelf-1 slot-2 3 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 2
admin> set line nfas-group-id = 1
admin> write
T1/{ shelf-1 slot-2 3 } written

admin> read t1 {1 2 4}
T1/{ shelf-1 slot-2 4 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 3
admin> set line nfas-group-id = 1
admin> write
T1/{ shelf-1 slot-2 4 } written
```

The following commands configure NFAS group 2, which contains lines 5 through 8:

```
admin> read t1 {1 2 5}
T1/{ shelf-1 slot-2 5 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 0
admin> set line nfas-group-id = 2
admin> set channel 24 channel = nfas-primary
admin> write
T1/{ shelf-1 slot-2 5 } written

admin> read t1 {1 2 6}
T1/{ shelf-1 slot-2 6 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 1
admin> set line nfas-group-id = 2
admin> set line channel 24 channel = nfas-secondary
```

```

admin> write
T1/{ shelf-1 slot-2 6 } written

admin> read t1 {1 2 7}
T1/{ shelf-1 slot-2 7 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 2
admin> set line nfas-group-id = 2
admin> write
T1/{ shelf-1 slot-2 7 } written

admin> read t1 {1 2 8}
T1/{ shelf-1 slot-2 8 } read

admin> set line signaling-mode = isdn-nfas
admin> set line nfas-id = 3
admin> set line nfas-group-id = 2
admin> write
T1/{ shelf-1 slot-2 8 } written

```

Configuring ISDN NFAS for Japanese switch types

To introduce non-facility associated signaling (NFAS) support for Japanese switches, TAOS unit supports implicit identification of the primary D-channel interface and explicit identification of all other interfaces, as required by Japanese switches.

Following is an example of PRI/T1 line configuration for NFAS with a Japanese switch:

```

admin> read t1 { 1 1 1}
T1/{ shelf-1 slot-1 1} read

admin> set line-interface signaling-mode = isdn-nfas
admin> set line-interface switch-type = japan-pri
admin> set line-interface nfas-group-id = 0
admin> set line-interface nfas-id = 0
admin> set line-interface channel 24 channel =
nfas-primary-d-channel

admin> write
T1/{ shelf-1 slot-1 1} written

```

Configuring T1 R1 and R1-Modified (Taiwan) with ANI and called-number processing

R1 is a multifrequency inband signaling system that uses a set of register signals known as MFR1 tones as addressing signals. Each address (telephone number) is preceded by a KP pulse and followed by an ST pulse denoting the end of addressing.

R1 signaling can optionally be used with Automatic Number Identification (ANI), which is similar to Caller ID (CLID). When it is in use, you can specify whether to send an Automatic Number ID Request (ANIR) to the switch. If you specify that the unit must send an ANIR to

Configuring T1 Cards

Configuring T1 R1 and R1-Modified (Taiwan) with ANI and called-number processing

the switch, you can also specify how long it waits before sending the request, and how long the ANIR signal lasts.

The following parameters enable R1 signaling on T1 lines and specify the timing of certain signals from the switch. These parameters are shown with their default settings:

```
[in T1/{ any-shelf any-slot 0 }:line-interface]
signaling-mode = inband
r1-use-anir = no
r1-first-digit-timer = 240
r1-anir-delay = 350
r1-anir-timer = 200
r1-modified = no
```

Parameter	Specifies
Signaling-Mode	For T1 R1 signaling, you must set Signaling-Mode to R1-Inband.
R1-Use-ANIR	Enables/disables ANI processing (CLID). It is set to No by default. If set to Yes, the system performs ANI processing on incoming calls.
R1-First-Digit-Timer	Time in milliseconds to wait for the first digit from the switch after sending the KP pulse. The default setting is 340 ms. The valid range is from 0 to 1000.
R1-ANIR-Delay	Time in milliseconds to wait before sending the ANIR signal after receipt of the ST pulse from the switch. The default setting is 350 ms. The valid range is from 300 to 2000.
R1-ANIR-Timer	Duration in milliseconds of the ANIR signal. The default setting is 200 ms. The valid range is from 180 to 400.
R1-Modified	Enables/disables a modified version R1 signaling that is required in Taiwan. It is set to No by default, which indicates regular R1 signaling (described in the ITU recommendation Q.310- 332). TAOS units located in Taiwan must set this parameter to Yes.

Following is an example that shows how to configure R1-Modified signaling (Taiwan) with ANIR in a T1 profile:

```
admin> read t1 { 1 5 1}
T1/{ shelf-1 slot-5 1 } read

admin> set line signal = r1-inband

admin> set line r1-use-anir = yes

admin> set line r1-first-digit-timer = 360

admin> set line r1-anir-delay = 360

admin> set line r1-anir-timer = 220

admin> set line r1-modified = yes

admin> write
T1/{ shelf-1 slot-5 1 } written
```


Configuring clocking

You can configure the TAOS unit to use any of the T1 lines as a master phase-locked loop (PLL) clock source for synchronous connections for an entire system. In synchronous transmission, both the sending device and the receiving device must maintain synchronization to determine where one block of data ends and the next begins.

From the T1 lines configured as eligible clock sources, the TAOS unit chooses a clock source on the basis of priority. If multiple T1 lines are configured as eligible clock sources and have an equal clock priority, the TAOS unit chooses one of them at random. Once chosen as the clock source, the line used until it becomes unavailable or a higher-priority source becomes available.

If no eligible external sources are available, the system uses an internal clock generated from the primary shelf controller. Using the internal clock is generally not recommended.

The Clock-Source diagnostic command displays the current master clock source and any available clock source. Sources from layer 2 up, which are preferred, are marked with an asterisk.

To specify a clock source and set a priority, proceed as follows after reading in the line's T1 profile:

```
admin> set clock-source = eligible
admin> set clock-priority = high-priority
admin> write
```

Configuring the front-end transceiver

The front-end type of the T1 transceiver can be CSU or DSX.

If you are connecting the TAOS unit to a DSX, set the Front-End-Type to DSX. With this setting you must also specify the length of the physical T1 line in feet. The value must reflect the longest line length you expect to encounter in your installation, up to a maximum of 655 feet (200m).

If you are not connecting the TAOS unit to a DSX, set Front-End-Type to CSU. You might also have to set a line buildout value to specify the amount of attenuation, in decibels, that the TAOS unit must apply to the line. If the TAOS unit is too close to a repeater, you need to add some attenuation to reduce the strength of the signal. Ask your service provider whether you need attenuation and, if so, how much.

To specify DSX settings, proceed as in the following example after reading in the line's T1 profile:

```
admin> set front-end-type = dsx
admin> set dsx-line-length = 1-133
admin> write
```

To specify CSU settings, proceed as in the following example after reading in the line's T1 profile:

```
admin> set front-end-type = csu
admin> set csu-build-out = 7.5-db
admin> write
```

Configuring channel usage

You must specify how each of the 24 channels of a T1 line is to be used. By default, T1 channels are configured as switched. (If you are going to set up the lines for NFAS, see “Configuring NFAS” on page 7-13 for additional channel-configuration information.)

You can configure each of the 24 channels of a T1 line for one of the following uses:

- **unused-channel**—Channel is unused. Send the single idle code defined for this channel.
- **switched-channel**—A switched channel, which will be robbed-bit or D channel, depending on how the line is configured at a higher level.
- **nailed-64-channel**—Clear-channel 64Kbps circuit. Does not require any setup information.
- **d-channel**—Channel is used for ISDN D channel signaling directed at the appropriate controller for the physical interface.
- **nfas-primary-d-channel**—The primary D channel for a group of T1 lines with the same NFAS ID. All other channels on the NFAS line must be set to **switched-channel**, **nailed-64-channel**, or **unused-channel**. Within an NFAS group, only one line should be configured to provide the primary ISDN D channel.
- **nfas-secondary-d-channel**—The secondary D channel for a group of T1 lines with the same NFAS ID. All other channels on the NFAS line must be set to **switched-channel**, **nailed-64-channel**, or **unused-channel**. Within an NFAS group, you configure only one line to provide the secondary (backup) D channel.

To specify the channel usage:

- 1 List the Line-Interface parameters:

```
admin> list line-interface
```

- 2 Set the Channel-Usage parameter for the first channel:

```
admin> set channel 1 channel-usage=[unused-channel |
switched-channel |nailed-64-channel| d-channel| nfas-pri-
mary-d-channel| nfas-secondary-d-channel]
admin> write
```

Assigning telephone numbers to switched channels

Channel assignments typically specify add-on numbers, not full telephone numbers. Add-on numbers include only the rightmost digits needed to distinguish one number from another. For example, if a line is assigned 23 numbers, all of which begin with 212-555-, the add-on number is the unique set of digits to the right of these common digits.

The most common reason multichannel calls fail to add channels properly is that the calling unit cannot use the add-on numbers it receives. To avoid this problem, make sure that the add-on numbers you assign all have the same number of digits.

When a caller initiates a multichannel call, it first dials the base channel and then requests additional numbers for dialing the additional channels. When it receives add-on numbers, the caller integrates them with the number it dialed for the base channel as follows:

- If the add-on number has fewer digits than the dialed number, the caller pads the add-on number with the leftmost digits that are included in the dialed number but not in the add-on number. For example, if the add-on number is 6532 and the dialed-number is 9-212-555-1212, the caller uses 9-212-555-6532 to dial the next channel.
- If the add-on number has more digits than the dialed number, the caller discards extra digits in the add-on numbers, starting with the leftmost digit.
- If the add-on number has the same number of digits as the dialed number, the entire add-on number is used. For example, if 6532 is the add-on number and 6588 is the dialed number, the caller uses 6532 to dial the next channel.

To assign add-on numbers to the channels of a T1 line, proceed as in the following example:

```
admin> list line channel
channel-config[1]={switched-channel 9 "" {any-shelf any-slot 0} 0 }
channel-config[2]={switched-channel 9 "" {any-shelf any-slot 0} 0 }
channel-config[3]={switched-channel 9 "" {any-shelf any-slot 0} 0 }
...
channel-config[24]={switched-channel 9 "" {any-shelf any-slot 0} 0 }
admin> set 1 phone = 60
admin> set 2 phone = 61
admin> set 3 phone = 62
admin> set 4 phone = 63
admin> set 5 phone = 64
```

In a hunt group, a group of channels is assigned the same telephone number. When a call comes in on that number, the TAOS unit uses the first available channel to which the number is assigned. Because channels in a hunt group share a common telephone number, the add-on numbers in the profile are all the same.

The following example shows how to configure two groups of four channels with hunt groups:

```
admin> set 6 phone = 70
admin> set 7 phone = 70
admin> set 8 phone = 70
admin> set 9 phone = 70
admin> set 10 phone = 72
admin> set 11 phone = 72
admin> set 12 phone = 72
```

```
admin> set 13 phone = 72
admin> write
```

Configuring trunk groups

Like nailed channels that have been assigned a group number, switched channels in a trunk group can be referred to from a Connection profile and Call-Route profile to direct outbound calls to use that specific bandwidth. Trunk groups also serve a variety of other purposes, such as separating lines supplied by different carriers so those lines can be used as backup for each other if one switch becomes unavailable. The decision to use trunk groups is a global one. Once you have enabled the use of trunk groups, *every* switched channel must be assigned a trunk group number or it will not be available for outbound calls.

Trunk groups limit the number of channels available to multichannel calls, because only channels within the same trunk group can be aggregated.

To enable trunk groups, open the System profile and set Use-Trunk-Groups to Yes, as in the following example:

```
admin> read system
SYSTEM read
admin> list
name = ""
system-rmt-mgmt = yes
use-trunk-groups = no
idle-logout = 0
parallel-dialing = 2
single-file-incoming = yes
analog-encoding = a-law
sessionid-base = 0
admin> set use-trunk-groups = yes
admin> write
```

Then assign the channels of each T1 line to a trunk group, as in the following example:

```
admin> list line channel 1
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set trunk-group = 4
admin> list .. 2
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set trunk-group = 4
admin> list .. 3
channel-usage = switched-channel
trunk-group = 9
```

```
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
```

```
admin> set trunk-group = 4
admin> write
```

Note: Command history is very useful for repeating commands. Press the Up-Arrow to redisplay the command, and then press Enter. (For more information, see the *TAOS Command-Line Interface Guide*.)

Configuring nailed channels

The number of nailed (leased) channels must be the same at both ends of the connection. For example, if there are five nailed channels at the local end, there must be five nailed channels at the remote end. However, channel assignments do not have to match. For example Channel 1 might be switched at the local end and nailed at the remote end.

Note that channels in a nailed group must be contiguous on the T1 line.

When you configure Connection profiles to use the leased connection, you must specify the Nailed-Group number in the Telco-Options subprofile.

To configure a nailed channel, proceed as in the following example:

```
admin> list line channel 1
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel = nailed
admin> set nailed = 3
admin> list .. 2
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel = nailed
admin> set nailed = 3
admin> write
```

Configuring a back-to-back T1 connection

For diagnostic purposes, you might sometimes want to configure a back-to-back T1 connection between ports on two TAOS units. In the T1 profile for one end of the line you want to connect with a back-to-back connection, specify the following values:

- Signaling-Mode set to Inband (the default)
- Robbed-Bit-Mode set to Wink-Start (the default)

Configuring T1 Cards

Specifying analog encoding for TAOS unit codecs

- Clock-Source set to Eligible (the default)

In T1 profile for the other end of the line, specify the following values:

- Signaling-Mode set to Inband (the default)
- Robbed-Bit-Mode set to Inc-W-200 or Inc-W-400
- Clock-Source set to Eligible (the default)

Connect the two ports with a T1-crossover cable. You can now configure Connection profiles between the units and dial over the connection as you would over the WAN. (For information about configuring Connection profiles, see the *APX 8000/MAX TNT/DSLNT WAN, Routing and Tunneling Configuration Guide*.)

Specifying analog encoding for TAOS unit codecs

Codecs connected to T1 use a different encoding standard for digitized analog data than do codecs connected to E1. The default for T1 is U-Law, the default for E1 is A-Law.

To specify the analog encoding, proceed as in the following example:

- 1 Open the System profile:

```
admin> read system
```

- 2 Specify the analog encoding for all the codecs in the TAOS unit:

```
admin> set analog-encoding = u-law
```

- 3 Write the System profile to save the changes:

```
admin> write  
SYSTEM written
```

Configuring specialized options

The settings described in this section are not normally used. Depending on your configuration, however, you might need to change the default values.

Typically, the D channel of a PRI line uses normal data. However, for some connections you might need to invert the data to avoid transmitting a pattern that the connection cannot handle. Inversion changes 1s to 0s and 0s to 1s. Both sides of the connection must agree to use inverted data.

Idle mode determines whether the D channel looks for a flag pattern (01111110) or a mark pattern (11111111) as the idle indicator. The default setting, Flag-Idle, is usually correct.

To set these options, use the Data-Sense and Idle-Mode parameters:

```
admin> set data-sense = [normal|inv]  
admin> set idle-mode = [mark-idle|flag-idle]  
admin> write
```

Sample T1 configuration

This section provides an example of how to configure a T1 slot card. The example uses the following setup:

- The card is in shelf 1, slot 2.
- All lines use PRI signaling.
- Switch type is NTI-PRI.
- The line is connected to a DSX and is less than 100 feet (30.5m) long. It therefore uses the default settings for Front-End-Type and DSX-Line-Length.
- All the channels are switched (the default), with the exception of channel 24, which is set for D channel signaling.
- All the channels are assigned to trunk group 9 (the default).
- The Default-Call-Type is digital (the default), so all calls received on this card are routed to the Hybrid Access (HDLC) card.
- The rest of the line parameters are left at their default values.

To configure the T1 card as in this example:

- 1 Create a new T1 profile:

```
admin> new t1
T1/{ any-shelf any-slot 0 } read
```

- 2 Set the physical address for the first T1 line:

```
admin> set physical-address ={ 1 2 1}
```

This applies the changes to the T1 line in the specified slot.

- 3 List the contents of the line profile:

```
admin> list line-interface
enabled = no
frame-type = d4
encoding = ami
clock-source = eligible
clock-priority = middle-priority
signaling-mode = inband
robbed-bit-mode = wink-start
default-call-type = digital
switch-type = att-pri
nfas-id = 0
call-by-call = 0
data-sense = normal
idle-mode = flag-idle
FDL = none
front-end-type = dsx
DSX-line-length = 1-133
CSU-build-out = 0-db
channel-config = [ { switched-channel 9 " { any-shelf
any-slot 0 } 0 } { switc+
maintenance-state = no
sendDisc-val = 0
```

- 4 Enable the line:

```
admin> set enabled = yes
```
- 5 Set the frame type:

```
admin> set frame-type = est
```
- 6 Set the line encoding:

```
admin> set encoding = b8zs
```
- 7 Set the signaling mode:

```
admin> set signaling-mode = isdn
```
- 8 Set the switch type:

```
admin> set switch-type = nti-pri
```
- 9 Next, assign all the channels to trunk group 7:

```
admin> set channel 1 truck-group = 7
```
- 10 Press the Up-Arrow key or Ctrl-P to redisplay the Set command you just entered.
- 11 Use the Left Arrow key or Control-B to change the channel number and trunk group for all the channels.
- 12 Change the channel usage of channel 24 to D Channel, because this channel carries the signaling for the PRI line.

```
admin> set channel 24 channel-usage = d -channel
```
- 13 Write the profile to commit your changes:

```
admin> write
```

```
T1/{ shelf-1 slot-2 2 } written
```
- 14 Because the T1 lines are all configured similarly, you can write the changes to the rest of the lines by setting the physical address and then writing the same profile for each of the lines:

```
admin> set physical-address = { 2 1 2}
admin> write
```

```
T1/{ shelf-1 slot-1 2 written
admin> set physical-address = { 2 1 3}
T1/{ shelf-1 slot-1 3} written
```

Continue until you have configured all the lines.

Default Call-Route profiles

When the TAOS unit detects that a T1 card has been installed, it creates on default Call-Route profile associated with the card. For example

```
admin> dir call-r
  9 12/11/1996 15:58:08 { { { any-shelf any-slot 0 } 0 } 0 }
 13 01/06/1997 17:17:10 { { { shelf-1 slot-2 0 } 0 }
```

This default Call-Route profile routes outbound trunk calls to any line on the card. To handle inbound modem and LAN-session traffic, you must configure specific call routes. For details, see Chapter 19, “Configuring Call Routing.”

Configuring T1 FrameLine Cards (MAX TNT, DSLTNT)

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Introduction to T1 FrameLine

The T1 FrameLine slot card provides 10 unchannelized T1 lines, each of which can be used for one nailed connection. Associated with each T1 line is a Serial Communications Adapter (SCA), which is responsible for receiving and transmitting HDLC frames. Because there is only one SCA per line, only one PPP or Frame Relay link (possibly with multiple DLCIs) can be active per line.

Unlike other slot cards, such as the Series56 II and Series56 III Digital Modems cards or Hybrid Access (HDLC) cards, call routing profiles are not used for the FrameLine card and are ignored if they exist. The data pathway is directed to an on-board SCA device and cannot be routed to another host card. All packetization of data occurs locally.

Overview of supported features

This section describes the T1 FrameLine slot card's support for the following protocols:

- PPP
- Frame Relay
- Routing protocols
- SNMP

PPP

The T1 FrameLine slot card supports PPP as follows:

- Only one PPP session per line.
- Bandwidth per session is 1-24 DS0 channels.
- Channels need not be contiguous.

- Multilink Protocol (MP) and Multilink Protocol Plus (MP+) are not supported. The connection profile must specify only PPP.
- Users are authenticated by the local profile or RADIUS.
- Stac compression is not supported.

Frame Relay

The T1 FrameLine slot card supports Frame Relay as follows:

- Only one Frame Relay link, possibly containing multiple data-link connection identifiers (DLCIs), can be active per line.
- Bandwidth per link is 1-24 DS0 channels.
- Channels need not be contiguous.
- Up to 240 permanent virtual circuits (PVCs) are supported per card.

Routing protocols

The T1 FrameLine slot card supports only IP routing.

RADIUS

The T1 FrameLine slot card supports the same RADIUS accounting and authentication as the digital modem cards.

SNMP

The T1 FrameLine slot card supports SNMP as follows:

- DS1 status and management are the same as for the eight-port T1 card.
- The T1 FrameLine slot card supports the accounting Management Information Base (MIB) for session information.

Overview of T1 FrameLine configuration

Configuring the T1 FrameLine slot card is similar to T1 slot card configuration except that the T1 FrameLine slot card has the following configuration restrictions:

- Signaling-Mode must be set to inband.
- The T1 FrameLine card can be used only for nailed Frame Relay or PPP links.
- You must set Channel-Usage to either Unused-Channel or Nailed-64-Channel.
- If Channel-Usage is Nailed-64-Channel and you are using nailed channels, the Nailed-Group setting must be unique to the line. Two different T1 lines cannot share a nailed group.
- Unlike the T1 card, channels in the same nailed group do not have to be contiguous. For example, DS0 channels 1 and 3 can be in the same nailed group, with channel 2 unused.
- The following T1 profile parameters are not applicable for the FrameLine card:

- Call-by-Call
- Channel-Usage
- Default-Call-Type
- Data Sense
- FDL
- Idle-Mode
- Maintenance-State
- NFAS-ID
- Robbed-Bit-Mode
- SendDisc-Val
- Switch-Type

For information about configuring T1 profiles, see Chapter 7, “Configuring T1 Cards.”

Configuring the clock source

The T1 FrameLine slot card uses the same system-wide PLL synchronous clock source for DS1 transmission as do the eight-port T1 and E1 cards. Any of the lines can serve as the clock source for the unit. To configure the T1 FrameLine card’s clock source, use the same parameters (Clock-Source and Clock-Priority) that you use for other cards.

All 10 lines must use the same clock source. Clocking on a per-line basis is not supported. The clock source can be one of the 10 lines, or a line on another slot card, or it can be internally generated from the primary shelf controller. Using the internal clock is not recommended. For more information about configuring the clock source, see “Configuring clocking” on page 7-17.

In addition, if the system clock source is from one of the 10 lines, it affects the timing on the Time-division multiplexing (TDM) backplane, because TDM timing is based on the clock source. This relationship exists even though the T1 FrameLine card does not use the TDM backplane.

Configuring E1 Cards

9

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Introduction to E1

An E1 line supports 32 64Kbps channels, each of which can be used to transmit and receive data or digitized voice. The line uses framing and signaling to achieve synchronous and reliable transmission. The most common configurations for E1 lines are PRI and unchannelized. (For information about provisioning your E1 line for use with the TAOS unit, see Appendix A, “Provisioning the Switch.”)

ISDN Primary Rate Interface (PRI)

In Europe, an E1/PRI line typically supports 30 B channels and one D channel. PRI configurations are used to receive multiple, simultaneous ISDN calls from analog-modem and digital-services dial-in traffic. Another common use of E1/PRI lines is to connect a private branch exchange (PBX) to a central office (CO) switch.

Nailed or unchannelized E1

An unchannelized E1 line can be used for nailed connections such as to a Frame Relay network. In such cases the configuration is static, and the TAOS unit treats the E1 line as if it were a single connection at a fixed speed, without individual channels.

Typically, when you pay your telephone company for a leased (nailed) line, you pay more for higher bandwidth. Anything in the range of 0bps to 2.048Mbps can be delivered on an E1 line, and provisioned at some 64Kbps fraction of the full E1 bandwidth.

Overview of E1 configuration

Table 9-1 lists the sections describing common tasks you might have to perform to configure an E1 line. The table includes a brief description of each task, and lists the parameters you will use.

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Table 9-1. E1 line configuration tasks

Section	Description of task	Associated parameters
“Understanding configuration requirements” on page 9-4	Before configuring your E1 line, gather the necessary information from your E1 service provider.	N/A
“Making a profile the working profile” on page 9-5	Before you can edit a profile, you must make it the working profile.	N/A
“Assigning names to E1 line profiles” on page 9-6	Assign a name to the E1 profile.	Name
“Enabling a line” on page 9-7	Make a line available for use.	Enabled

Table 9-1. E1 line configuration tasks (continued)

Section	Description of task	Associated parameters
“Configuring a back-to-back connection” on page 9-7	A back-to-back connection lets you connect two TAOS units to one another over a crossover E1 cable.	Back-to-Back
“Specifying the framing” on page 9-7	Framing specifies how the bits are sent on the line.	Frame-Type
“Specifying E1 signaling” on page 9-8	Specify the type of signaling used for your E1 line.	Signaling-Mode
“Configuring ISDN PRI signaling” on page 9-8	You must specify the type of network switch providing ISDN service on an E1 PRI line.	Switch-Type
“Configuring ISDN network-side emulation” on page 9-9	ISDN emulation enables you to build, send, receive, and process ISDN data.	ISDN-Emulation-Side
“Configuring E1 R1 signaling” on page 9-10	R1 is a multifrequency inband signaling protocol that uses a set of register signals known as MFR1 tones as addressing signals.	Signaling-Mode Switch-Type
“Configuring E1 R2 signaling” on page 9-10	Specify R2 signaling and specify R2-specific configuration options.	Signaling-Mode Number-Complete Group-B-Signal Group-II-Signal Answer-Delay
“Configuring DPNSS signaling” on page 9-12	Specify Digital Private Network Signaling System (DPNSS) signaling and associated options.	Signaling-Mode Layer3-End Layer2-End NL-Value Loop-Avoidance
“Configuring overlap receiving on PRI lines” on page 9-13	T1 or E1 PRI lines with overlap receiving enable the TAOS unit to gather the complete called number from the network switch via a series of Information messages, enabling the use of features such as called-number authentication.	Signaling-Mode Overlap-Receiving PRI-Prefix-Number Trailing-Digits T302-Timer
“Configuring clocking” on page 9-13	Set Clock-Source to specify whether the E1 line can be used as the master clock source for synchronous connections. Also specify the priority of the E1 lines to be used for clocking.	Clock-Source Clock-Priority

Table 9-1. E1 line configuration tasks (continued)

Section	Description of task	Associated parameters
“Configuring the front-end E1 transceiver” on page 9-13	Set the front end type of the E1 transceiver to Long-Haul or Short-Haul, depending on the type of termination your line uses.	Front-End-Type
“Configuring channel usage” on page 9-14	Specify how each of the E1 channels is to be used.	Channel-Usage
“Assigning telephone numbers to switched channels” on page 9-14	Typically, you specify only the rightmost digits needed to distinguish one number from another. These are called add-on numbers.	Phone-Number
“Configuring trunk groups” on page 9-14	A trunk group is a group of channels that has been assigned a number.	Trunk-Group
“Configuring nailed channels” on page 9-15	You must assign a nailed channel to a group to make it available for use. The group number can be referred to in a Connection or Frame-Relay profile to specify a permanent leased connection using that group of nailed channels.	Nailed-Group
Chapter 19, “Configuring Call Routing”	<p>The TAOS unit uses call routing to determine where to route incoming and outgoing calls. The preferred way to set up call-routing is to put all call routing information in one place: a Call-Route profile.</p> <p>If you do not use Call-Route profiles, specify the physical address of a device to which calls received on this channel are routed.</p>	Default-Call-Type Call-by-Call-Service Shelf Slot Item

Understanding configuration requirements

You need the following information from your E1/PRI service provider:

- The telephone numbers assigned to your E1/PRI interface, channel-by-channel
- Nailed-up channels (also called private WAN), if any
- Unused channels, if any
- Switch type (or emulation)—DPNSS only
- Switch layers 2 and 3 configuration—Digital Access Signaling System (DASS) 2 and DPNSS only (A/B end, X/Y end)

- Rate adaption protocol—DASS 2 and DPNSS only (X.30 and V.110)

Note: The TAOS unit cannot receive multichannel calls using Multilink Protocol (MP) encapsulation unless all channels of the call share a common telephone number (namely, a hunt group). You can request that your service provider supply you with a hunt group.

Making a profile the working profile

When the TAOS unit system detects that an E1 card has been installed, it creates a default E1 profile for each of the eight lines on the card.

In the following display example, the Dir command shows eight default E1 profiles created for a card installed in slot 2:

```
admin> dir e1
305 12/11/1996 15:58:20 { shelf-1 slot-2 2 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 4 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 5 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 6 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 7 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 8 }
320 12/20/1996 20:55:31 { shelf-1 slot-2 3 }
317 01/08/1997 09:58:55 { shelf-1 slot-2 1 }
```

By default, a line is not enabled, which means that it is not available for use. Its default signaling method is inband, typically used for channelized connections.

To configure an E1 profile, make it the working profile by reading it into the edit buffer. For example:

```
admin> read e1 {1 2 1}
E1/{ shelf-1 slot-2 1 } read
```

Once you have read in a profile, it remains the working profile until you read in another profile. You can use the Set command to change one or more of the profile's parameters.

To save your configuration changes, use the Write command. For example,

```
admin> write
E1/{ shelf-1 slot-2 1 } written
```

To list the parameters in an E1 profile, use the List command, as in the following example:

```
admin> list
physical-address* = { shelf-1 slot-2 1 }
line-interface = { yes esf b8zs eligible middle-priority
isdn wink-star+
```

The following example shows the parameters in an E1 profile, with sample settings:

```
[in E1/{ shelf-1 slot-15 5 }]
name = ""
physical-address* = { shelf-1 slot-15 5 }
```

```
line-interface = { no none g703 eligible middle-priority +
back-to-back = false

[in E1/{ shelf-1 slot-15 5 }:line-interface]
enabled = yes
t-online-type = none
frame-type = g703
clock-source = eligible
clock-priority = middle-priority
signaling-mode = isdn
default-call-type = digital
switch-type = net5-pri
incoming-call-handling = reject-all
front-end-type = short-haul
overlap-receiving = no
pri-prefix-number = ""
trailing-digits = 2
t302-timer = 10000
channel-config = [ { unused-channel 9 "" { any-shelf +
layer3-end = x-side
layer2-end = b-side
nl-value = 64
loop-avoidance = 7
number-complete = end-of-pulsing
group-b-answer-signal = signal-b-6
group-b-busy-signal = signal-b-3
group-ii-signal = signal-ii-2
input-sample-count = one-sample
answer-delay = 200
caller-id = no-caller-id
hunt-grp-phone-number-1 = ""
hunt-grp-phone-number-2 = ""
hunt-grp-phone-number-3 = ""
collect-incoming-digits = no
r1-use-anir = no
r1-first-digit-timer = 340
r1-anir-delay = 350
r1-anir-timer = 200
r1-modified = no

[in E1/{ shelf-1 slot-15 5 }:line-interface:channel-con +
channel-usage = unused-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 1
```

Assigning names to E1 line profiles

In an E1 profile, the Name parameter enables you to assign the profile a name. The name can include up to 16 characters. After you assign it, it is displayed after the line's physical address in the Dir command output. For example:

```
admin> read e1 {1 12 0}
admin> set name = E1 Trunk
admin> write
E1/{ shelf-1 slot-12 0 } written
admin> dir e1
17  04/17/1997 19:00:02  { shelf-1 slot-12 0 }  "E1 Trunk"
```

For E1 lines, the Line Status window displays either the name (if assigned) or the physical address. If the name is longer than eight characters, the last character displayed is a plus sign (+).

Enabling a line

By default each E1 line is disabled. To enable an E1 line, read its profile to make it the working profile, then set Enabled to Yes, as in the following example:

```
admin> read e1 {1 2 1}
E1/{ shelf-1 slot-2 1 } read
admin> set line enabled = yes
admin> write
E1/{ shelf-1 slot-2 1 } written
```

Configuring a back-to-back connection

For diagnostics, you can configure DASS-2 or DPNSS lines in a back-to-back connection. A crossover cable connects an E1 port of one TAOS unit to an E1 port of another TAOS unit. No switch is required, and the connection is entirely local. One TAOS unit must be set up for data terminal operation (DTE) operation, and the other for data circuit-terminating equipment (DCE) operation.

To specify a back-to-back connection, set the Back-to-Back parameter in the E1 profile:

```
admin> read e1 {1 2 1}
E1/{ shelf-1 slot-2 1 } read
admin> set back-to-back = [true|false]
admin> write
E1/{ shelf-1 slot-2 1 } written
```

Specifying the framing

The E1 framing mode can be G703 (G.704 with CRC4, the standard framing mode used by most E1 ISDN and DASS2 providers) or 2DS (G.704 without CRC4, a variant of G.703 required by most E1 DPNSS providers in the United Kingdom). If the line is not configured for ISDN signaling, you can use the D4 format, also known as superframe.

Your E1 service provider must provide the correct framing values for your lines.

To specify the framing, set the Frame-Type parameter:

```
admin> read e1 {1 2 1}
E1/{ shelf-1 slot-2 1 } read
```

```
admin> set line frame-type = [G703|2DS|D4|esf]
admin> write
E1/{ shelf-1 slot-2 1 } written
```

Specifying E1 signaling

An E1 line's signaling mode can be any of the following:

- ISDN
- DPNSS (DPNSS or DASS 2 signaling)
- Channel-associated signaling (CAS). CAS signaling modes includes the following:
 - E1-R2-Signaling (R2 signaling)
 - E1-Argentina-Signaling
 - E1-Brazil-Signaling
 - E1-Chinese-Signaling (R2 signaling used in China)
 - E1-Czech-Signaling
 - E1-India-Signaling
 - E1-Korean-Signaling (R2 signaling used in Korea)
 - E1-Malaysia-Signaling
 - E1-Metered-Signaling (metered R2 signaling, used in Brazil and South Africa)
 - E1-Philippine-Signaling
 - E1-P7-Signaling (R2 P7)
 - R1-Inband

In the E1 profile Line-Interface subprofile, configure E1 signaling as follows:

```
admin> read e1 {1 2 1}
E1/{ shelf-1 slot-2 1 } read
admin> set line signaling-mode = signalingmode
admin> write
E1/{ shelf-1 slot-2 1 } written
```

Replace *signalingmode* with one of the modes listed above. If you are using one of the CAS signaling modes, you must also set the Switch-Type parameter to CAS.

For more information on the E1 signaling parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Configuring ISDN PRI signaling

For ISDN signaling you must also specify the type of switch providing E1/PRI service to your TAOS unit. Obtain the information from your ISDN carrier.

When you set the signaling mode to ISDN, you must also set channel 17 as the D channel. Note that ISDN signaling often requires ESF framing and B8ZS encoding.

Configure ISDN PRI service as in the following example:

```
admin>read e1 {1 15 5}
E1/{ shelf-1 slot-15 5 } read

admin>list
[in E1/{ shelf-1 slot-15 5 }]
name = ""
physical-address* = { shelf-1 slot-15 5 }
line-interface = { no none g703 eligible middle-priority
isdn +
back-to-back = false

admin>list line-interface
[in E1/{ shelf-1 slot-15 5 }:line-interface]
enabled = no
t-online-type = none
frame-type = g703
clock-source = eligible
clock-priority = middle-priority
signaling-mode = isdn
default-call-type = digital
switch-type = net5-pri
..
..

admin> set frame-type = esf
admin> set signaling-mode = isdn
admin> set switch-type = switchtype
admin> set channel 17 channel-usage=d-channel
admin> write
```

To see a complete list of switch types supported on the TAOS unit, see the TAOS unit command-line interface online help or the *APX 8000/MAX TNT/DSLNT Reference*.

Configuring ISDN network-side emulation

You can configure PRI lines to use either network-side or user-side ISDN emulation. Previously, PRI lines on the TAOS unit supported only user-side emulation. Following is the relevant parameter, shown with its default setting:

```
[in E1/{ any-shelf any-slot0 }:line-interface]
isdn-emulation-side = te
```

ISDN is a nonsymmetrical protocol used by telephone carriers to provide digital services to end users. There are no ISDN links between telephone carrier Central Offices (COs). ISDN links exist only between the CO and the customer. Therefore, an ISDN link can be viewed as having two sides—the network side, or network terminating (NT) equipment, and the user side, or terminal equipment (TE). The user side can connect only to the network side, and vice versa. Both the network side and the user side perform the same functions, but the format of the messages is different. For example, the network side must always set a bit and the user side must always clear it. These differences allow either side to determine whether the other end is the right one.

ISDN emulation enables you to build, send, receive, and process ISDN data. ISDN monitoring, on the other hand, allows you only to decode the ISDN data.

Configuring E1 R1 signaling

R1 is a multifrequency inband signaling protocol that uses a set of register signals known as MFR1 tones as addressing signals. Each address (telephone number) is preceded by a KP pulse and followed by an ST pulse denoting the end of addressing.

The R2 signaling option must be software licensed (hash-code enabled) on the system for R1 signaling to work. If one or more E1 lines on an E1 card are configured for R1 signaling, no other line on the card can use R2 signaling.

Following are the parameters relevant to R1 signaling, shown with sample values:

```
[in E1/{shelf-1 slot-13 1}:line-interface]
signaling-mode = r1-inband
switch-type = cas
```

All other line signaling parameters can be left in their default settings. The following example specifies R1 signaling on an E1 line in shelf 1, slot 13:

```
admin> read e1 {1 13 1}
E1/{ shelf-1 slot-13 1 } read

admin> set line signaling-mode = r1-inband

admin> set switch-type = cas

admin> write
E1/{ shelf-1 slot-13 1 } written
```

Configuring E1 R2 signaling

R2 signaling is an ITU-T standardized signaling protocol, which can be used on E1 digital trunks for switched circuits. It uses a combination of A/B bit manipulation in channel 16 of the E1 frame (line signaling), and inband MF tone generation and detection (register signaling). The relevant specifications are in ITU-T recommendations Q.400 to Q.490. R2 signaling is widely implemented in international markets where ISDN PRI is not yet available. The default bandwidth for data calls coming in over E1 channels using R2 signaling is 64Kbps.

To configure R2 signaling, you might need to set some or all of the following parameters:

Parameter	Specifies
Switch-Type	Type of switch the TAOS unit connects to. For R2 signaling, you must set Switch-Type to Switch-CAS. When the line is configured for channel associated signaling (CAS), the TAOS unit does not receive bearer-capability information from the carrier. Therefore, it cannot determine whether a call is voice-service or digital-service. For call-routing purposes, all calls on inband lines are assumed to be digital calls.
Answer-Delay	Milliseconds the TAOS unit delays before answering an R2 call.

Parameter	Specifies
Number-Complete	Number of digits considered to be a complete number on an incoming call using R2 signaling. You can specify End-of-Pulsing to have the TAOS unit continue receiving digits until the caller stops sending them, or you can specify a fixed number of digits (up to 10). In all cases, the digits received before the call is answered are considered the called number for call-routing purposes.
Group-B-Answer-Signal	Replaces the Group-B-Signal parameter found in earlier releases. It specifies the group-B signal that the TAOS unit sends before answering a call, and can be set to a value from Signal-B-1 to Signal-B-15. The default is Signal-B-6, which is the recommended setting for E1_R2 Israeli signaling.
Group-B-Busy-Signal	Group-B-Busy-Signal specifies the group-B signal that the TAOS unit sends as a busy signal. When the TAOS unit does not have sufficient resources to handle the call correctly (for example, if all of its modems are busy), it sends the group-B signal specified by this parameter. It can be set to a value from Signal-B-1 to Signal-B-15. The default is Signal-B-3, which is the recommended setting for E1_R2 Israeli signaling.
Group-II-Signaling	Group II signal that is sent in the course of an outgoing call, immediately after acknowledgment by the called end that all necessary address digits have been received. It is used for outgoing call configuration.
Caller-ID	<p>Enables or disables the use of caller ID for R2 calls. You must specify one the following signaling modes to enable the TAOS unit to process CLID information received from the switch:</p> <ul style="list-style-type: none"> • E1-Argentina-Signaling • E1-Brazil-Signaling • E1-Chinese-Signaling • E1-India-Signaling • E1-Israel-Signaling • E1-Kuwait-Signaling • E1-Malaysia-Signaling • E1-Mexico-Signaling • E1-New-Zealand-signaling • E1-Philippine-Signaling • E1-Thailand-Signaling <p>For details about configuring CLID authentication in a Connection profile, see the <i>APX 8000/MAX TNT/DSLNT WAN, Routing, and Tunneling Configuration Guide</i>.</p>

To configure the line for R2 signaling, proceed as in the following example:

```
admin> read e1 {1 2 2}
E1/{ shelf-1 slot-2 2 } read
```

```
admin> list line
enabled=no
frame-type=g703
clock-source=eligible
clock-priority=middle-priority
signaling-mode=isdn
switch-type=net5-pri
front-end-type=short-haul
channel-config=[ { unused-channel 9 "" { any-shelf any-slot+
..
..
admin> set line enabled = yes
admin> set line frame-type = 2DS
admin> set line signaling-mode = e1-r2-signaling
admin> set line switch-type = switch-cas
admin> set line number-complete = end-of-pulsing
admin> set line group-b-signal = signal-b-6
admin> set line group-ii-signal = signal-ii-2
admin> set line answer-delay = 200
admin> set line caller-id = get-caller-id
admin> write
E1/{ shelf-1 slot-2 2 } written
```

Configuring DPNSS signaling

When you are connecting to a DASS 2 or DPNSS switch, you must set the following parameters:

- Layer3-End specifies CCITT Layer 3. It must be set to X-Side (its default value).
- Layer2-End specifies CCITT Layer 2. It must be set to B-Side (its default value).
- NL-Value must be set to 64 (its default value).
- Loop-avoidance must be set to 7 (its default value).

Contact the service provider for more details. (These settings are not required for ISDN signaling.)

In the following example, an administrator configures DPNSS signaling using a Mercury switch (a variant of DPNSS). The specified framing mode, 2DS, is a variant of G.703 required by most E1 DPNSS providers in the United Kingdom. To configure an E1 line for DPNSS signaling, proceed as in the following example:

```
admin> read e1 {1 2 2}
E1/{ shelf-1 slot-2 2 } read
admin> set enabled = yes
admin> set signaling-mode = e1-dpnss-signaling
admin> set switch = mercury-dpnss
admin> set frame-type = 2ds
admin> set layer3-end = x-side
admin> set layer2-end = b-side
admin> set nl-value = 64
```



```
admin> set loop-avoidance = 7
admin> write
```

Configuring overlap receiving on PRI lines

Overlap receiving affects the procedure of establishing an incoming call received on a T1 or E1 PRI line in the TAOS unit. With overlap receiving, the TAOS unit can gather the complete called number from the network switch via a series of Information messages, enabling the use of features such as called-number authentication. For information about configuring overlap receiving on PRI lines, see Chapter 7, “Configuring T1 Cards.”

Configuring clocking

You can configure the TAOS unit to use any of the E1 lines as a master clock source for synchronous connections for an entire system. In synchronous transmission, both the sending device and the receiving device must maintain synchronization in order to determine where one block of data ends and the next begins.

From the E1 lines configured as eligible clock sources, the TAOS unit chooses a clock source on the basis of priority. If multiple E1 lines are configured as eligible clock sources and have an equal clock priority, the TAOS unit chooses one of them at random. Once chosen as the clock source, the line is used until it becomes unavailable or a higher-priority source becomes available.

If no eligible external sources are available, the system uses an internal clock generated from the primary shelf controller. Using the internal clock is generally not recommended.

The Clock-Source diagnostic command displays the current master clock source. Enter the command on the shelf controller to display which slot (if any) is being used as the clock source. Enter the command on an E1 card to display which line is used.

To specify a clock source and set a priority, proceed as follows after reading in the line's E1 profile:

```
admin> set clock-source = eligible
admin> set clock-priority = high-priority
admin> write
```

Configuring the front-end E1 transceiver

The front-end type of the E1 transceiver can be short haul or long haul. Long haul is only for lines using 120-ohm termination.

Specify the front-end settings as follows, after reading in the line's E1 profile:

```
admin> set front-end-type=[ short-haul | long-haul ]
admin> write
```

Configuring channel usage

You must specify how each of the 32 channels of an E1 line is to be used. By default, E1 channels are configured as switched. Each of the 32 channels of an E1 line can be configured for one of the following uses:

- Unused-Channel—Channel is unused. Send the single idle code defined for this channel.
- Switched-Channel—Switched channel, using either robbed-bit or D-channel signaling, depending on how the line is configured at a higher level.
- Nailed-64-Channel—A clear-channel 64Kbps circuit. This configuration does not require any setup information.
- D-Channel—Channel 16 (channel 17 in the TAOS unit interface) is used for ISDN D-channel signaling directed at the appropriate controller for the physical interface.

To specify the channel usage:

- 1 List the Line-Interface parameters:

```
admin> list line-interface
```

- 2 Set the Channel-Usage parameter for the first channel:

```
admin> set channel 1 channel-usage=[unused-channel|switched-channel|nailed-64-channel|d-channel]
admin> write
```

Assigning telephone numbers to switched channels

Assigning telephone numbers to switched E1 channels is no different from assigning them to switched T1 channels. See “Assigning telephone numbers to switched channels” on page 7-19.

Configuring trunk groups

Like nailed channels that have been assigned a group number, switched channels in a trunk group can be referred to from a Connection profile and Call-Route profile to direct outbound calls to use that specific bandwidth. Trunk groups also serve a variety of other purposes, such as separating lines supplied by different carriers so those lines can be used as backup for each other if one switch becomes unavailable. The decision to use trunk groups is a global one. Once you have enabled the use of trunk groups, *every* switched channel must be assigned a trunk group number or it will not be available for outbound calls.

Trunk groups limit the number of channels available to multichannel calls, because only channels within the same trunk group can be aggregated.

To enable trunk groups, open the System profile and set Use-Trunk-Groups to Yes, as in the following example:

```
admin> read system
SYSTEM read
admin> list
name = ""
system-rmt-mgmt = yes
use-trunk-groups = no
```

```
idle-logout = 0
parallel-dialing = 2
single-file-incoming = yes
analog-encoding = a-law
sessionid-base = 0
admin> set use-trunk-groups = yes
admin> write
```

Then assign the channels of each E1 line to a trunk group, as in the following example:

```
admin> read e1 {1 1 1}
E1/{ shelf-1 slot-1 1 } read

admin> list line channel 1
[in E1/{ shelf-1 slot-15 1 }:line-interface:channel-con +]
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set trunk-group = 4

admin> list .. 2
[in E1/{ shelf-1 slot-15 1 }:line-interface:channel-con +]
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set trunk-group = 4

admin> list .. 3
[in E1/{ shelf-1 slot-15 1 }:line-interface:channel-con +]
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set trunk-group = 4

..
..
admin> write
```

Note: Command history is very useful for repeating commands. Press the Up Arrow key to redisplay the command, and then press Enter. (For more information, see the *TAOS Command-Line Interface Guide*.)

Configuring nailed channels

The number of nailed (leased) channels must be the same at both ends of the connection. For example, if there are five nailed channels at the local end, there must be five nailed channels at the remote end. However, channel assignments do not have to match. For example, Channel 1 can be switched at the local end and nailed at the remote end. Channels in a nailed group must be contiguous on the E1 line.

When you configure Connection profiles to use the leased connection, you must specify the Nailed-Group number in the Telco-Options subprofile.

To configure a nailed channel, proceed as in the following example:

```
admin> list line channel 1
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel = nailed
admin> set nailed = 3
admin> list .. 2
channel-usage = switched-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel = nailed
admin> set nailed = 3
admin> write
```

Specifying analog encoding for TAOS unit codecs

Codecs connected to T1 use a different encoding standard for digitized analog data than do codecs connected to E1. The default for T1 is U-Law, the default for E1 is A-Law.

To specify the analog encoding, proceed as in the following example:

- 1 Open the System profile:

```
admin> read system
```
- 2 Specify the analog encoding for all the codecs in the TAOS unit:

```
admin> set analog-encoding = a-law
```
- 3 Write the System profile to save the changes:

```
admin> write
SYSTEM written
```

Default Call-Route profiles

When the TAOS unit detects that an E1 card has been installed, it creates one default Call-Route profile associated with the card. For example:

```
admin> dir call-r
  9  12/11/1996 15:58:08  { { { any-shelf any-slot 0 } 0 }
0 }
 13  01/06/1997 17:17:10  { { { shelf-1 slot-2 0 } 0 } 0 }
```

This default Call-Route profile routes outbound trunk calls to any line on the card. To handle inbound modem and LAN-session traffic, you must configure specific call routes. For details, see Chapter 19, “Configuring Call Routing.”

Configuring E1 FrameLine Cards (MAX TNT, DSLTNT)

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Introduction to E1 FrameLine

The E1 FrameLine slot card provides 10 E1 FrameLine lines, each of which can be used for one nailed connection. Associated with each E1 line is a Serial Communications Adapter (SCA), which is responsible for receiving and transmitting HDLC frames. Because there is only one SCA per line, only one PPP or Frame Relay link (possibly with multiple DLCIs) can be active per line.

Unlike other slot cards, such as the Series56 II and Series56 III Digital Modems cards or Hybrid Access (HDLC) cards, call routing profiles are not used for the E1 FrameLine card and are ignored if they exist. The data pathway is directed to an on-board SCA device and cannot be routed to another host card. All packetization of data occurs locally.

Overview of supported features

This section describes the E1 FrameLine slot card's support for the following protocols:

- PPP
- Frame Relay
- Routing protocols
- SNMP

PPP

The E1 FrameLine slot card supports PPP as follows:

- Only one PPP session per line.
- Bandwidth per session is 1-31 DS0 channels. Channel 1 is not available.
- Channels need not be contiguous.
- Multilink Protocol (MP) and Multilink Protocol Plus (MP+) are not supported. The connection profile must specify only PPP.
- Users are authenticated by the local profile or RADIUS.
- Stac compression is not supported.

Frame Relay

The E1 FrameLine slot card supports Frame Relay as follows:

- Only one Frame Relay link, possibly containing multiple data-link connection identifiers (DLCIs), can be active per line.
- Bandwidth per link is 1-31 DS0 channels. Channel 1 is not available.
- Channels need not be contiguous.
- Up to 120 PVCs are supported per card.

Routing protocols

The E1 FrameLine slot card supports only IP routing.

RADIUS

The E1 FrameLine slot card supports the same RADIUS accounting and authentication as the digital modem cards.

SNMP

The E1 FrameLine slot card supports SNMP as follows:

- DS1 status and management are the same as for the eight-port E1 card.
- The E1 FrameLine slot card supports the accounting Management Information Base (MIB) for session information.

Overview of E1 FrameLine configuration

Configuring the E1 FrameLine slot card is similar to E1 slot card configuration except that the E1 FrameLine slot card has the following configuration restrictions:

- Signaling-mode must be set to E1-No-Signaling.
- Frame-Type must be set to G703.
- T-Online-Type must be set to None.

- Channel-Usage for channel must be set to Unused-Channel.
- For all other channels, Channel-Usage must be set to either Unused-Channel or Nailed-64-Channel.
- Channel 17 is usable.
- You cannot have the same nailed group on two different E1 lines.
- Unlike the E1 card, channels in the same nailed group do not need to be contiguous. For example, channels 1 and 3 can be in same nailed group with channel 2 unused.
- Only the following E1 profile parameters are applicable for the E1 FrameLine slot card:
 - Enabled
 - T-Online-Type
 - Frame-Type
 - Clock Source
 - Signaling-Mode
 - Channel-Usage
 - Nailed-Group

For complete information on configuring E1 lines, refer to Chapter 8, “Configuring E1 Cards.”

Example E1 FrameLine configuration

When you install the E1 FrameLine slot card, the MAX TNT or DSLTNT creates 10 E1 profiles. The following is the default line-interface configuration:

```
enabled = no
t-online-type = none
frame-type = g703
clock-source = eligible
clock-priority = low-priority
signaling-mode = e1-no-signaling
default-call-type = digital
switch-type = net5-pri
front-end-type = short-haul
overlap-receiving = no
pri-prefix-number = ""
trailing-digits = 2
t302-timer = 10000
layer3-end = x-side
layer2-end = b-side
nl-value = 64
loop-avoidance = 7
number-complete = end-of-pulsing
group-b-answer-signal = signal-b-6
group-b-busy-signal = signal-b-3
group-ii-signal = signal-ii-2
answer-delay = 200
caller-id = no-caller-id
hunt-grp-phone-number-1 = ""
```

Configuring E1 FrameLine Cards (MAX TNT, DSLTNT)

Administrative profiles for E1 FrameLine

```
hunt-grp-phone-number-2 = ""
hunt-grp-phone-number-3 = ""
```

To configure the E1 FrameLine card:

```
admin> read E1 {1 2 2}
UE1/{ shelf-1 slot-2 2 } read
admin> set enabled = yes
admin> list channel 1
channel-usage = unused-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel-usage = nailed-64-channel
admin> set nailed-group = 3
admin> list .. 2
channel-usage = unused-channel
trunk-group = 9
phone-number = ""
call-route-info = { any-shelf any-slot 0 }
nailed-group = 0
admin> set channel-usage = nailed-64-channel
admin> set nailed-group = 3
Continue configuring the rest of the channels similarly. When you have finished, write the
profile:
admin> write
UE1/{ shelf-1 slot-2 2 } written
```

Administrative profiles for E1 FrameLine

In addition to the E1 profile described in the previous section, the following administrative profiles apply to the E1 FrameLine slot card:

- Admin-State profile
- Device-State profile
- Slot-Info profile
- T1-Status profile

This section explains the changes to these profiles to support the E1 FrameLine card.

Admin-State profile

When you install the E1 FrameLine slot card, the MAX TNT or DSLTNT creates 20 Admin-State profiles; 10 are associated with the E1 lines and 10 are associated with the SCA devices that do HDLC framing. Profiles are retained during card resets. The unit deletes these profiles if you install a different type of card into a slot. You can also delete the profiles using the Slot command with the `-r` option.

The profile index is displayed as `{ shelf slot N }`

- An *N* value of 1-10 identifies a E1 line on the card.
- An *N* value of 11-20 identifies an SCA on the card.
- An SCA value of 11 is associated with line 1, an SCA of 12 with line 2, and so on.

Device-State profile

The TNT or DSLTNT creates a Device-State profile for each DS0 and each SCA when the E1 FrameLine slot card enters the up state.

You use the DS0-related profiles as you do the eight-port E1 slot card profiles.

You use the SCA related profiles as you do the Series56 II and III Digital Modem cards *except* that setting the Req'd-State parameter to Down-Req'd-State when a call is active on that SCA has no effect.

The profile index is `{ { shelf slot N } M }`

- An *N* value of 1-10 identifies a line on the card.
- An *N* value of 11-20 identifies an SCA on the card.
- An *M* value is the DS0 channel number. Its range is [1..32] for E1. For an SCA, *M* is always 0.

Administrative commands and status information

You can maintain the E1 FrameLine slot card as you do the eight-port E1 card:

- The Dircode and Show commands display the E1 FrameLine loads as `10-unchan-E1-card`.
- You can view the status of the SCAs with the HDLC command.
- The line status is identical to the line status for the eight-port E1 card.
- You can view the errors on each line by opening a session to the card and using the E1-Stats command.

For more information about diagnostics on the E1 FrameLine card, see the *APX 8000/MAX TNT/DSLNT Administration Guide*.

Configuring the clock source

The E1 FrameLine slot card uses the same system-wide PLL synchronous clock source for DS1 transmission as do the eight-port T1 and E1 cards. Any of the lines can serve as the clock source for the unit. To configure the E1 FrameLine card's clock source, use the same parameters (Clock-Source and Clock-Priority) that you use for other cards.

All 10 lines must use the same clock source. Clocking on a per-line basis is not supported. The clock source can be one of the 10 lines, or a line on another slot card, or it can be internally generated from the primary shelf controller. Using the internal clock is generally not recommended.

In addition, if the system clock source is from one of the 10 lines, it affects the timing on the TDM backplane, because TDM timing is based on the clock source. This relationship exists even though the E1 FrameLine slot card does not use the TDM backplane.

Configuring T3 Cards

11

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Introduction to T3

The T3 slot card is a communications circuit composed of seven DS2s, each of which includes four DS1s, each of which in turn is composed of 24 DS0s, for a total of 672 DS0 channels.

On the T3 card, DS2 channel 1 includes DS1 lines 1-4, DS2 channel 2 includes DS1 lines 5-8, and so on. Each DS1 is similar to a T1 line, except that on the T3 card, a DS1 functions only if the DS2 and DS3 of which it is a component are operating and in frame.

You can think of the T3 card as 28 T1 lines, because it provides 28 independently configurable DS1 lines. Each of the DS1 lines has the same capabilities as the DS1 lines on a T1 card. Use of SNMP for DS1-level management and status monitoring of the T3 card is the same as for the eight-port T1 card. No SNMP or status monitoring is currently available at the DS3 level.

Overview of T3 configuration

Table 11-1 lists the sections describing common tasks you might have to perform to configure a T3 line. The table includes a brief description of each task and lists the parameters you will use.

(This chapter describes only the specifics that apply to a T3 card. For information about configuring T1 profiles, see Chapter 7, “Configuring T1 Cards.”)

Configuring T3 Cards

Understanding T3 configuration requirements

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSL/TNT Reference*.

Table 11-1. T3 line configuration tasks

Section	Description of task	Associated parameters
“Understanding T3 configuration requirements” on page 11-2	Although you configure the T3 card similarly to the eight-port T1 card, there are important differences you must understand before configuring the card.	Clock-Source Clock-Priority NFAS-ID FDL Front-End-Type DSX-Line-Length CSU-Buildout
“Understanding T3 slot card profiles” on page 11-3	The TAOS unit creates a single T3 profile and 28 T1 profiles for each T3 card in the system.	N/A
“Assigning a name to a T3 profile” on page 11-4	Assign a name to the T3 profile.	Name
“Enabling a line” on page 11-5	Make a line available for use.	Enabled
“Configuring the T3 physical link” on page 11-5	Before you configure the T1 profiles that make up the T3 card, you must first configure the T3 physical line parameters in the T3 profile.	Physical-Address Enabled Frame-Type Line-Length
“Configuring clocking” on page 11-5	Any of the T1 lines associated with a T3 card can be configured as the clock source for the TAOS unit.	Clock-Source Clock-Priority

Understanding T3 configuration requirements

Configuring the T3 slot card is very similar to configuring the eight-port T1 slot card, but with some important differences. Table 11-2 lists the differences.

Table 11-2. Differences between T3 card configuration and T1 card configuration

Parameter	Difference
NFAS-ID	The T3 card supports up to 14 NFAS groups. An NFAS group can be composed of up to 28 lines, subject to the limitations of the switch. NFAS is configured in the same way as for the eight-port T1 card.

Table 11-2. Differences between T3 card configuration and T1 card configuration (continued)

Parameter	Difference
FDL	The DS1-level FDL services supported by the T3 card are the same as for the eight-port T1 card. DS3-level FDL capabilities such as the Far-End Alarm and Control Channel (FEAC) and Path Maintenance Data Link are currently unsupported. (For information on specifying FDL, see the <i>APX 8000/MAX TNT/DSLNT Administration Guide</i> .)
Front-End-Type DSX-Line-Length CSU-Build-Out	These parameters are ignored in T1 profiles that apply to the T3 card.

Understanding T3 slot card profiles

When the TAOS unit first detects the presence of a T3 slot card, it creates the following profiles for each card:

- One T3 profile
- One Call-Route profile
- 28 T1 profiles (one for each DS1 on the T3 card)

T3 profile

When the TAOS unit first detects the presence of a T3 card, it creates a default T3 profile for the card. For example, after installing a T3 card installed in slot 7, you can verify the creation of a T3 profile as follows:

```
admin> dir t3
7 03/21/1997 21:12:03 { shelf-1 slot-7 0 }
```

The following example shows the parameters in a T3 profile, with sample settings:

```
t3 { shelf-N slot-N N }
  physical-address* = { shelf-N slot-N N }
  enabled = yes
  application = m13
  line-length = 1-255
```

Call-Route profile

At the same time that it creates a T3 profile, the TAOS unit creates one default Call-Route profile that routes outbound trunk calls to any line on the card. You can display the Call-Route profile as shown in the following example:

```
admin> dir call-r
9 02/28/1997 10:54:38 { { { any-shelf any-slot 0 } 0 } 0 }
```

```
13 02/28/1997 10:54:49 { { { shelf-1 slot-8 0 } 0 } 0 }
13 02/28/1997 10:54:49 { { { shelf-1 slot-11 0 } 0 } 0 }
13 02/28/1997 10:54:49 { { { shelf-1 slot-16 0 } 0 } 0 }
13 02/28/1997 10:54:49 { { { shelf-1 slot-13 0 } 0 } 0 }
13 03/21/1997 10:18:40 { { { shelf-1 slot-7 0 } 0 } 0 }
```

T1 profiles

The TAOS unit also creates 28 T1 profiles for the T3 interface. You use these profiles to configure parameters for each of the DS1s that make up the T3.

The following example shows the parameters in a T1 profile, with sample settings:

```
T1 { shelf-N slot-N N }
    name=
    physical-address* = { shelf-N slot-N N }
    line-interface
        enabled = no
        frame-type = d4
        encoding = ami
        clock-source = eligible
        clock-priority = middle-priority
        signaling-mode = inband
        robbed-bit-mode = wink-start
        default-call-type = digital
        switch-type = att-pri
        nfas-id = 0
        call-by-call = 0
        data-sense = normal
        idle-mode = flag-idle
        FDL = none
        front-end-type = dsx
        DSX-line-length = 1-133
        CSU-build-out = 0-db
        maintenance-state = no
    channel-config N
        channel-usage = switched-channel
        trunk-group = 9
        phone-number = ""
        call-route-info = { any-shelf any-slot 0 }
        nailed-group = 0
```

These T1 profiles are identical to those created for the DS1s on an eight-port T1 card.

Assigning a name to a T3 profile

In a T3 profile, the Name parameter enables you to assign the profile a name of up to 16 characters. It is displayed after the line's physical address in the Dir command output. For example:


```
admin> read t3 {1 12 0}
admin> set name = T3 Trunk
admin> write
T3/{ shelf-1 slot-12 0 } written
admin> dir T3
17 04/17/1997 19:00:02 { shelf-1 slot-12 0 } "T3 Trunk"
```

For T3 lines, the Line Status window displays the first eight characters of the name if one has been assigned. For example:

```
"T3 Trunk" 1/15/00 LA la la la la la la
```

If the name is longer than eight characters, the last character displayed is a plus sign (+).

Enabling a line

By default each DS3 line is disabled. When the DS3 interface is disabled, it transmits the DS3 Idle Signal to the far end.

To enable a T3 line, read its profile to make it the working profile, then set the Line-Interface subprofile's Enabled parameter to Yes, as in the following example:

```
admin> read t3 {1 2 1}
T3/{ shelf-1 slot-2 1 } read
admin> set enabled = yes
admin> write
T3/{ shelf-1 slot-2 1 } written
```

Configuring the T3 physical link

You must specify a frame type and the length of the lines that connect the TAOS unit T3 slot card to the DSX-3 cross-connect. The line length must reflect the longest line length you expect to encounter in your installation. For a direct connection, double the value.

To configure the T3 card's physical link, read its profile into the edit buffer, and enter the following commands:

```
admin> set frame-type = [m13|c-bit-parity]
admin> set line-length = [0-225|226-450]
admin> write
```

Configuring clocking

For DS1 transmission, the T3 slot card uses the same system-wide PLL synchronous clock source used by the eight-port T1 cards. Any of the T3 card's T1 lines can serve as the clock source for the TAOS unit system.

(For information about specifying a clock source, see "Configuring clocking" on page 7-17.)

Configuring Serial WAN (SWAN) Cards (MAX TNT, DSLTNT)

12

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Introduction to SWAN

The Serial WAN (SWAN) card, has four V.35 serial ports, which can be used for nailed Frame Relay connections. This card can support up to 120 Frame Relay virtual circuits. A serial WAN port provides a V.35/RS-449 WAN interface that is typically used for connecting to a Frame Relay switch. The clock speed received from the link determines the serial WAN data rate. The maximum acceptable speed is 8 Mbps. The clock speed at the serial WAN port has no effect on the bandwidth of other WAN interfaces in the MAX TNT or DSLTNT unit.

Overview of SWAN configuration

Table 12-1 lists the sections describing common tasks you might have to perform to configure a SWAN line. The table includes a brief description of each task and lists the parameters you will use.

Configuring Serial WAN (SWAN) Cards (MAX TNT, DSLTNT)

Understanding SWAN card configuration requirements

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Table 12-1. SWAN-card configuration tasks

Section	Description of task	Associated parameters
“Understanding SWAN card configuration requirements” on page 12-2	Explains important configuration information you should understand before you configure the SWAN card.	N/A
“Making a profile the working profile” on page 12-3	Before you can edit a profile, you must make it the working profile.	N/A
“Assigning a name to a SWAN profile” on page 12-4	Assign a name to the SWAN profile.	Name
“Enabling a line” on page 12-4	Make a line available for use.	Enabled
“Specifying a nailed group” on page 12-4	The nailed group is used to assign a Frame Relay connection to a SWAN line.	Nailed-Group
“Specifying the SWAN internal clock speed” on page 12-5	The SWAN slot card can generate a transmit internal clock based on the clock speed of its Serial Communication Adapter (SCA) chips.	Clock-Mode Divider Exp

Understanding SWAN card configuration requirements

Table 12-2 provides important configuration information you might need before configuring your SWAN card.

Table 12-2. SWAN card configuration

Element	Explanation
Connections	The SWAN card currently supports only nailed Frame Relay connections.
Call routing information	Call routing information for the SWAN card is currently ignored.
Trunk groups	Trunk groups are not currently implemented for the SWAN card.

Table 12-2. SWAN card configuration (continued)

Element	Explanation
Activation	<p>The Activation parameter tells the MAX which signals control the data flow through the serial WAN port. The DCE to which the serial WAN port is connected (for example, a Frame Relay switch) determines how to set the serial WAN port Activation value. Flow control is always handled by the Clear To Send (CTS) signal.</p> <p>Currently, the Activation parameter supports only one value: Static.</p>

Making a profile the working profile

When the TAOS unit detects that a SWAN card has been installed, it creates a default SWAN profile for each of the lines on the card.

In the following example, the Dir command displays default SWAN line profiles created for a card installed in slot 2:

```
admin> dir SWAN
305 12/11/1996 15:58:20 { shelf-1 slot-2 1 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 2 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 3 }
305 12/11/1996 15:58:20 { shelf-1 slot-2 4 }
```

By default, the line is not enabled, which means that it is not available for use. Its default signaling method is inband, typically used for channelized connections.

To list the parameters in a SWAN profile, use the List command, as in the following example:

```
admin> list
name = " "
physical-address* = { any-shelf any-slot 0 }
enabled = no
line-config = { 0 0 static { any-shelf any-slot 0 } }
```

Following is an example of a SWAN profile, with its parameters set to sample values:

```
SWAN { shelf-N slot-N N }
name = 1:14:2
physical-address* = { shelf-1 slot-14 2 }
enabled = no
line-config
  trunk-group = 0
  nailed-group = 2
  activation = static
  call-route-info
    shelf = any-shelf
    slot = any-slot
    item-number = 0
```

```
clocking
  clock-mode = external-clock
  divider = 1
  exp = 2
```

Assigning a name to a SWAN profile

In a SWAN profile, the Name parameter enables you to assign the profile a name of up to 16 characters. By default, the name displays the address of the card as *shelf:slot:item*. Note that the TAOS unit uses only the physical address to identify the SWAN line.

The name is displayed after the line's physical address in the Dir command output. For example:

```
admin> read SWAN {1 12 0}
admin> set name = SWAN1
admin> write
SWAN/{ shelf-1 slot-12 0 } written
admin> dir SWAN
17  04/17/1997 19:00:02 { shelf-1 slot-12 0 }  "SWAN1"
```

For SWAN lines, the Line Status window displays the first eight characters of the name if one has been assigned. If the name is longer than eight characters, the last character displayed is a plus sign (+).

Enabling a line

By default each SWAN line is disabled. To enable a SWAN line, read its profile to make it the working profile, then set Enabled to Yes, as in the following example:

```
admin> read SWAN {1 2 1}
SWAN/{ shelf-1 slot-2 1 } read
admin> set enabled = yes
admin> write
SWAN/{ shelf-1 slot-2 1 } written
```

Specifying a nailed group

The Nailed-Group parameter assigns a nailed group number to the SWAN line. The setting, which must also be specified in a Frame-Relay profile, directs the Frame Relay connection to use this line.

To specify a nailed group, proceed as in the following example:

```
admin> read SWAN {1 2 1}
SWAN/{ shelf-1 slot-2 1 } read
admin> set line nailed-group= 5
admin> write
SWAN/{ shelf-1 slot-2 1 } written
```

Specifying the SWAN internal clock speed

The SWAN slot card can generate a transmit or receive internal clock based on the clock speed of its Serial Communication Adapter (SCA) chips. The maximum clock speed is 5.55 MHz.

To generate an internal clock for a SWAN line, you configure the following parameters:

Parameter	Description
Clock-Mode	Specifies whether the SWAN card generates an internal clock. External-Clock (the default) specifies the SWAN line receives clock from an external source. Internal-Clock specifies the SWAN line generates its own clock. If set to External-Clock, none of the other parameters in the Clocking profile apply.
Divider	The number by which the SCA internal clock speed, 16.667 MHz, is divided to calculate the internal clock speed. Valid values are from 1 to 256.
Exp	The exponent which is used to calculate the internal clock speed. Valid values are from 0 to 9.

The SWAN card uses the following formula to generate its internal clock:

$$\text{clock speed (MHz)} = (16.667 / \text{divider}) / (2 \text{ to the } \text{exp power})$$

The following example shows how to configure an internally generated clock speed:

- 1 Read the SWAN profile:

```
admin>read swan {1 13 2}
SWAN/{ shelf-1 slot-13 2 } read
```

- 2 List the profile:

```
admin>list
[in SWAN/{ shelf-1 slot-13 2 }]
name = 1:13:2
physical-address* = { shelf-1 slot-13 2 }
enabled = yes
line-config = { 0 61 static { any-shelf any-slot 0 } { exte +
```

- 3 List the Line-Config profile

```
admin>list line-config
[in SWAN/{ shelf-1 slot-13 2 }:line-config]
trunk-group = 0
nailed-group = 61
activation = static
call-route-info = { any-shelf any-slot 0 }
clocking = { external-clock 1 2 }
```

- 4 List the Clocking subprofile:

```
admin>list clocking
[in SWAN/{ shelf-1 slot-13 2 }:line-config:clocking]
clock-mode = external-clock
```

```
divider = 1
```

```
exp = 2
```

- 5 Specify the Divider and exponent to use for calculating the clock speed:

```
admin>set divider=4
```

```
admin>set exp=2
```

- 6 Write the profile:

```
admin>write
```

This example sets the internally generated clock to 1.042 Mhz—that is, $(16.667/4)/2^2=1.042$.

Frame Relay configuration

Complete details about Frame Relay configuration can be found in the *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide*.

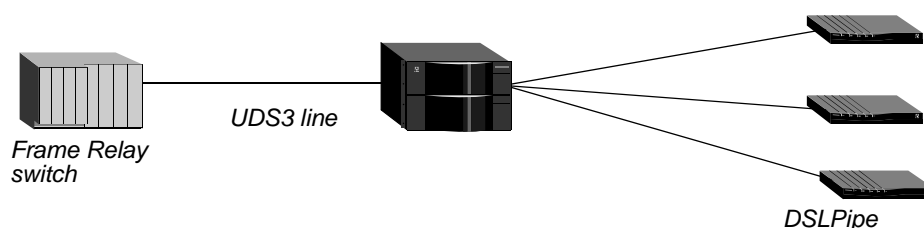
Configuring Unchannelized DS3 Cards (MAX TNT, DSLTNT)

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Introduction to unchannelized DS3

The unchannelized DS3 slot card (UDS3), supported on the MAX TNT and the DSLTNT, is a 44.736Mbps communications circuit that can be used to concentrate incoming traffic on the unit and direct it to a Frame Relay switch. Figure 9-1 shows an example of an unchannelized DS3 slot card application.

Figure 13-1. Example of unchannelized DS3 slot card application



Supported features

The unchannelized DS3 slot card (UDS3) provides support for the following:

- One Frame Relay link per line, possibly containing multiple DLCIs
- IP and IPX routing
- Layer 2 frame relay switching
- The DS3 MIB (RFC 1407)

Overview of unchannelized DS3 configuration

Table 9-1 lists the sections describing common tasks you might have to perform to configure an unchannelized DS3 line. The table includes a brief description of each task, and lists the parameters you will use.

For complete information about the associated parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Table 13-1. Unchannelized DS3 line configuration tasks

Task	Description	Associated parameters
“Using the UDS3 profile” on page 13-2	The unit creates a single unchannelized DS3 profile when you install the unchannelized DS3 slot card, which you use to configure the card.	N/A
“Configuring the UDS3 physical link” on page 13-2	Assign a name and enable the unchannelized DS3 line.	Name Enabled

Using the UDS3 profile

When the unit first detects the presence of an unchannelized DS3 slot card, it creates a default (UDS3) profile for the card. For example, after installing an unchannelized DS3 slot card in slot 7, you can verify the creation of a UDS3 profile as follows:

```
admin> dir uds3
7 03/21/1997 21:12:03 { shelf-1 slot-7 0 }
```

The following example shows the parameters in a UDS3 profile, with sample settings:

```
admin> read uds3 { 1 7 1 }
UDS3/{ shelf-1 slot-7 1 } read
admin> list
name = 1:7:1
physical-address* = { shelf-1 slot-7 1 }
enabled = yes
line-config = { 0 131 static { any-shelf any-slot 0 } c-bit-parity+
  trunk-group = 0
  nailed-group = 131
  activation = static
  call-route-info = { any-shelf any-slot 0 }
  line-type = c-bit-parity
  line-coding = b3zs
  loopback = no-loopback
```

Configuring the UDS3 physical link

In an unchannelized DS3 (UDS3) profile, the Name parameter enables you to assign the profile a name of up to 16 characters. It is displayed after the line’s physical address in the Dir command output.

By default, each unchannelized DS3 line is disabled. When the DS3 interface is disabled, it transmits the DS3 Idle Signal to the far end.

To assign the line a name and enable it, proceed as in the following example:

```
admin> read uds3 {1 2 1}
UDS3/{ shelf-1 slot-2 1 } read
admin> set name = uds3-LA
admin> set enabled = yes
admin> write
UDS3/{ shelf-1 slot-2 1 } written
```

The default settings for the `line-type` and `line-coding` parameters are used because the unchannelized DS3 slot card (UDS3) supports only C-bit-parity framing and B3ZS encoding.

Consult the *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide* for detailed information about configuring the Frame Relay portion of the connection.

Configuring DS3-ATM Cards

14

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Overview of DS3-ATM settings	14-1
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Introduction DS3-ATM

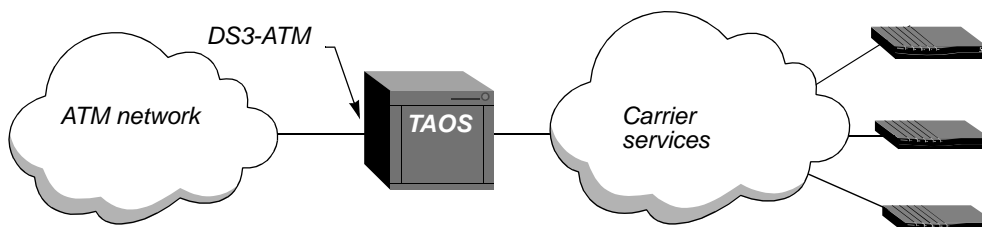
The DS3-ATM cards (DS3-ATM and DS3-ATM2) support one 44.736Mbps interface for connecting to one ATM switch. At a minimum, you must enable the line and specify a nailed group. TAOS units use the nailed group to direct traffic to the interface.

Note: The DS3-ATM card is supported on the MAX TNT and DSLTNT platforms only. The DS3-ATM2 card is supported on the APX 8000, MAX TNT, and DSLTNT platforms. Both cards use the same configuration profiles, but at the time of this writing, the DS3-ATM2 card does not support ATM-Frame Relay circuit configurations. Maximum performance with the DS3-ATM2 card is achieved using RFC 1483 ATM-AAL5-CPCS-PDU encapsulation.

This chapter refers to both the DS3-ATM and DS3-ATM2 cards as *DS3-ATM cards*.

An example use of a DS3-ATM interface is shown in Figure 14-1.

Figure 14-1. DS3-ATM interface to ATM network



Overview of DS3-ATM settings

A TAOS unit creates a DS3-ATM profile for each DS3-ATM interface detected in the system. Following are the relevant parameters, shown with default values:

```
[in DS3-ATM/{ any-shelf any-slot 0 }]
name = ""
physical-address* = { any-shelf any-slot 0 }
enabled = no
```

```
[in DS3-ATM/{ any-shelf any-slot 0 }:line-config]
trunk-group = 9
nailed-group = 1
activation = static
call-route-info = { any-shelf any-slot 0 }
loopback = no-loopback
high-tx-output = no
framer-mode = C-BIT-PLCP
vpi-vci-range = 0-15/32-4095
traffic-shapers = [ { no 1000 1000 2 no 0 } { no 1000 1000 2 no 1 } {+
cell-payload-scramble = yes
```

Parameter	Specifies
Name	Name of the interface. The default value is the interface address in <i>shelf:slot:item</i> format (for example, 1:2:3), but you can assign a text string of up to 16 characters.
Physical-Address	Physical address of the DS3-ATM port in the TAOS unit.
Enabled	Enable/disable the DS3-ATM interface. (Disabled by default.) When the interface is disabled, it transmits the DS3 Idle Signal to the far end.
Trunk-Group	<i>Not currently used.</i> Leave the default value (9).
Nailed-Group	Nailed-group number for the DS3-ATM physical interface. A Connection or RADIUS profile specifies this number to make use of the interface. Each interface must be assigned a number from 1 to 1024 that is unique within the system.
Activation	Line activation mode. Only the <code>static</code> setting is currently supported.
Call-Route-Info	<i>Not currently used.</i> Leave the default value (the zero address).
Loopback	Enable/disable loopback for diagnosing connectivity or possible equipment problems. Loopback is disabled by default, which is required for normal operations. For details, see “Looping back the line” on page 14-4.
High-Tx-Output	Enable/disable high transmit output. The default is <code>no</code> , which is correct for DS3-ATM cables that are less than 255 feet (78m) long. For cables longer than 255 feet, set this parameter to <code>yes</code> .
Framer-Mode	DS3-ATM framer mode. Valid values are <code>C-bit-ADM</code> and <code>C-BIT-PLCP</code> (the default). You can specify C-bit Physical Layer Convergence Protocol (PLCP) or C-bit ATM Direct Mapping (ADM) framing format for a DS3-ATM interface. Both sides of a DS3-ATM link must agree about the framing format. The PLCP format incurs some overhead for framing. ADM format does not. When ADM framing is used, the entire DS3 payload is used for ATM cells.
VPI-VCI-Range	Valid range of VCI numbers to be used with specified VPIs for virtual channel connections (VCCs). For details about setting the VPI-VCI range, see the <i>APX 8000/MAX TNT/DSLNTNT ATM Configuration Guide</i> .

Parameter	Specifies
Traffic-Shapers	Settings for shaping traffic on the interface. For details, see the <i>APX 8000/MAX TNT/DSLNT ATM Configuration Guide</i> .
Cell-Payload-Scramble	Enable/disable scrambling and descrambling of the 48-byte ATM cell payload. This function is enabled by default. Disable it only if the far-end switch has disabled the corresponding functions.

Examples of DS3-ATM configurations

The following set of commands enables a DS3-ATM interface in slot 12 and assigns the nailed group number 111.

```
admin> read ds3-atm {1 12 1}
DS3-ATM/{ shelf-1 12 1 } read

admin> set enabled = yes

admin> set line-config nailed-group = 111

admin> write
DS3-ATM/{ shelf-1 12 1 } written
```

Configuring redundant cards

When you are using two cards in a redundant configuration (as described in the hardware installation guide for your unit), both cards must use the same nailed group number to enable profiles to transparently use either card. After installing and cabling redundant cards, configure the primary card. For example:

```
admin> read ds3-atm {1 2 1}
DS3-ATM/{ shelf-1 2 1 } read

admin> set enabled = yes

admin> set line-config nailed-group = 100

admin> write
DS3-ATM/{ shelf-1 2 1 } written
```

Next, configure the secondary card with the same nailed group. For example:

```
admin> read ds3-atm {1 3 1}
DS3-ATM/{ shelf-1 3 1 } read

admin> set enabled = yes

admin> set line-config nailed-group = 100

admin> write
DS3-ATM/{ shelf-1 3 1 } written
```

If the primary card goes down, the TAOS unit switches over to the secondary card and reestablishes the link.

Looping back the line

To diagnose possible line problems, you can loop back the DS3-ATM interface by using the Loopback parameter in the line profile. While the interface is looped back, normal data traffic is interrupted. The Loopback parameter in the DS3-ATM profile supports the following settings:

- No-Loopback. The default, specifies that the line is operating normally.
- Facility-Loopback. During a facility loopback, the card returns the signal it receives on the line. The remote end receives back the signal it transmitted.
- Local-Loopback. During a local loopback, the receive path is connected to the transmit path at the DS3 multiplexer, enabling the slot card to receive what it transmitted.

For example, the following commands activate a local loopback:

```
admin> read ds3-atm {1 3 1}
DS3-ATM/{ shelf-1 slot-3 1 } read
admin> set line loopback= local-loopback
admin> write
DS3-ATM/{ shelf-1 slot-3 1 } written
```

To end the loopback, set the Loopback parameter to No-Loopback. For example:

```
admin> set line loopback = no-loopback
admin> write
DS3-ATM/{ shelf-1 slot-3 1 } written
```

For more details about checking line status and performing line checks, see the *APX 8000/MAX TNT/DSLNT Administration Guide*.

Configuring OC3-ATM Cards (MAX TNT/DSLNT)

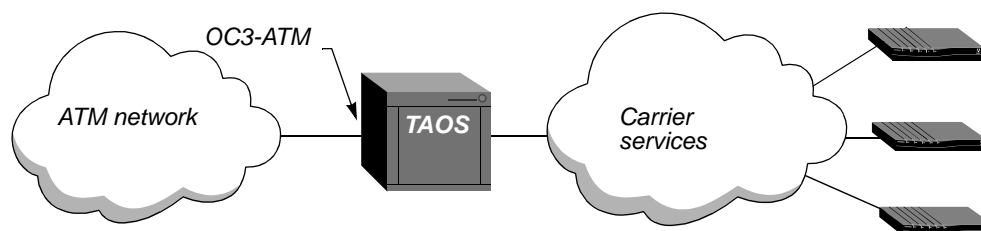
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Example of an OC3-ATM configuration	15-4

Introduction to OC3-ATM

An OC3-ATM card provides one 155.52Mbps interface for connecting to an ATM switch. At a minimum, you must enable the line and specify a nailed group. MAX TNT and DSLNT units use the nailed group to direct traffic to the interface.

An example use of an OC3-ATM interface is shown in Figure 15-1.

Figure 15-1. OC3-ATM interface to ATM network



Overview of OC3-ATM settings

A MAX TNT or DSLNT unit creates an OC3-ATM profile for each OC3-ATM interface detected in the system. Following are the relevant parameters, shown with default values:

```
[in OC3-ATM/{ any-shelf any-slot 0 } (new)]
name = ""
physical-address* = { any-shelf any-slot 0 }
enabled = no

[in OC3-ATM/{ any-shelf any-slot 0 }:line-config (new)]
trunk-group = 0
nailed-group = 1
call-route-info = { any-shelf any-slot 0 }
loopback = no-loopback
framer-mode = sdh
framer-rate = STS-3c
```

Configuring OC3-ATM Cards (MAX TNT/DSL TNT)

Overview of OC3-ATM settings

```
rx-descramble-disabled = no
tx-scramble-disabled = no
rx-cell-payload-descramble-disabled = no
tx-cell-payload-scramble-disabled = no
loop-timing = yes
vpi-vci-range = 0-15/32-4095
clock-source = not-eligible
clock-priority = middle-priority
traffic-shapers = [ { no 1000 1000 2 no 0 } { no 1000 1000 2 no 1 } { +
```

Parameter	Specifies
Name	Name of the interface. The default value is the interface address in <i>shelf:slot:item</i> format (for example, 1:2:3), but you can assign a text string of up to 16 characters.
Physical-Address	Physical address of the OC3-ATM port in the TAOS unit unit.
Enabled	Enable/disable the OC3-ATM interface. (Disabled by default.) When the OC3-ATM interface is disabled, it transmits the OC3-ATM Idle Signal to the far end.
Trunk-Group	<i>Not currently used.</i> Leave the default value (zero).
Nailed-Group	Nailed-group number for the OC3-ATM physical interface. A Connection or RADIUS profile specifies this number to make use of the interface. Each interface must be assigned a number from 1 to 1024 that is unique within the system.
Call-Route-Info	<i>Not currently used.</i> Leave the default value (the zero address).
Loopback	Enable/disable loopback for diagnosing connectivity or possible equipment problems. Loopback is disabled by default, which is required for normal operations.
Framer-Mode	Frame format for data transmitted on the interface. Valid settings are <i>sdh</i> (the default) and <i>sonet</i> , which represent the synchronous digital hierarchy (SDH) and synchronous optical network (SONET) frame formats, respectively.
Framer-Rate	Framing operations. Only the default STS-3C setting is used, which represents both the 155.52Mbps interface in the U.S. and the equivalent European 155.52Mbps interface (STM-1).
Rx-Descramble-Disabled	Enable/disable descrambling of the entire receive stream. This function is enabled by default. Disable it by setting this parameter to <i>yes</i> only if the far-end switch has disabled the corresponding functions.
Tx-Scramble-Disabled	Enable/disable scrambling of the entire transmit stream. This function is enabled by default. Disable it by setting this parameter to <i>yes</i> only if the far-end switch has disabled the corresponding functions.
Rx-Cell-Payload-Descramble-Disabled	Enable/disable descrambling of the 48-byte ATM cell payload in received cells. This function is enabled by default. Disable it by setting this parameter to <i>yes</i> only if the far-end switch has disabled the corresponding functions.

Parameter	Specifies
Tx-Cell-Payload-Scramble-Disabled	Enable/disable scrambling of the 48-byte ATM cell payload in transmitted cells. This function is enabled by default. Disable it by setting this parameter to <code>yes</code> only if the far-end switch has disabled the corresponding functions.
Loop-Timing	Enable/disable derivation of transmission timing from receiver inputs. Loop timing is enabled by default. If the parameter is set to <code>No</code> , transmission timing is derived from the reference clock instead.
VPI-VCI-Range	Valid range of VCI numbers to be used with specified VPIs for virtual channel connections (VCCs). For details about setting the VPI-VCI range, see the <i>APX 8000/MAX TNT/DSLNT ATM Configuration Guide</i> .
Clock-Source	Enable/disable obtaining the system clock signal from the port. By default, ports are not eligible clock sources. For information about using an OC3-ATM line as the source for the system clock, see “Using OC3-ATM ports as a clock source” on page 15-3.
Clock-Priority	Priority of the interface as the system’s clock source: high, middle, or low priority. Once the TAOS unit chooses a clock source, it uses that source until the interface becomes unavailable or a higher-priority source becomes available.
Traffic-Shapers	Settings for shaping traffic on the interface. For details, see the <i>APX 8000/MAX TNT/DSLNT ATM Configuration Guide</i> .
Tx-Scramble-Disabled	Enable/disable scrambling and descrambling of the entire transmit and receive stream. This function is enabled by default. Disable it only if the far-end switch has disabled the corresponding functions.
Tx-Cell-Payload-Scramble-Disabled	Enable/disable scrambling and descrambling of the 48-byte ATM cell payload in transmitted and received cells. This function is enabled by default. Disable it only if the far end switch has disabled the corresponding functions.

Using OC3-ATM ports as a clock source

OC3-ATM profiles support Clock-Source and Clock-Priority parameters for specifying whether the port can be used to source the ATM network clock signal and feed it to the shelf controller as the master clock for the unit. Each OC3-ATM port can be configured as eligible or ineligible for this use, and can be assigned a high, middle, or low priority for being elected as the clock source.

If more than one line is eligible to be the clock source, the system chooses the one with the highest priority, as specified by the Clock-Priority setting. If multiple sources of equal priority are present, the system selects the first valid clock source. (A clock source is valid if the Clock-Source parameter is set to `eligible` and the OC3-ATM interface is synchronized.)

Once it has selected a clock source, the system uses that source until the source becomes unavailable or a higher-priority source becomes available. If no eligible external sources are available, the system uses an internal clock generated by the shelf controller.

Configuring OC3-ATM Cards (MAX TNT/DSL TNT)

Example of an OC3-ATM configuration

For example, the following commands configure an OC3-ATM port as an eligible clock source. If this port becomes unavailable and is not backed up, the unit begins using the built-in clock on the shelf controller.

```
admin> read oc3-atm { 1 7 1 }
OC3-ATM/{ shelf-1 slot-7 1 } read

admin> set line-config clock-source = eligible

admin> set line-config clock-priority = high

admin> write
OC3-ATM/{ shelf-1 slot-7 1 } written
```

Example of an OC3-ATM configuration

In this example, the administrator enables an OC3-ATM interface in slot 7, assigns the nailed group number 222 to the interface, and sets the VPI-VCI range to allow an 8-bit VPI number:

```
admin> read oc3-atm {1 7 1}
OC3-ATM/{ shelf-1 slot-7 1 } read

admin> set enabled = yes

admin> set line-config nailed-group = 222

admin> set line-config vpi-vci-range = 0-255/32-255

admin> write
OC3-ATM/{ shelf-1 slot-7 1 } writtens
```

Note: For details about configuring ATM features and connections, see the *APX 8000/MAX TNT/DSL TNT ATM Configuration Guide*.

Configuring STM-0 Cards

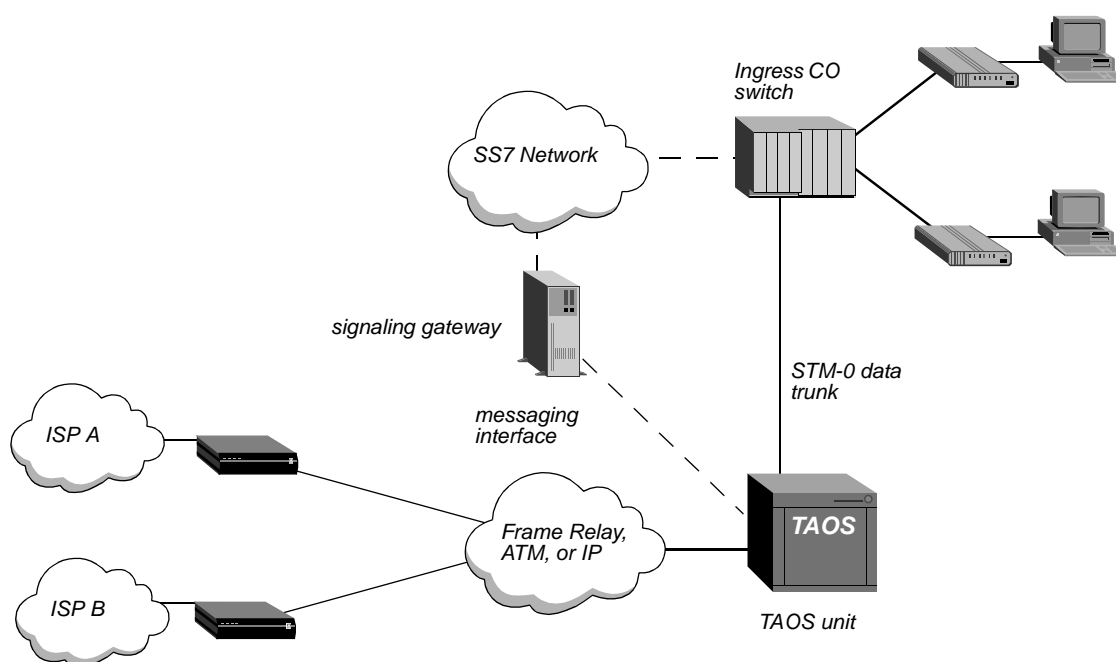
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Introduction to STM-0

The Synchronous Transport Module (STM)-0 card is an optical 51.85Mbps communication circuit designed to be used with an approved signaling gateway. Each of its 28 T1 lines can be configured as Signaling System 7 (SS7) data trunks. When configured as an SS7 data trunk, the signaling gateway takes control of the data trunks, instructing the TAOS unit when to bring calls up or down.

Note that the STM-0 card does not support Call-Routing profiles, PRI signaling, or inband signaling. Figure 16-1 shows an example STM-0 configuration.

Figure 16-1. Example STM-0 configuration



Using STM and T1 profiles

When the TAOS unit first detects the presence of an STM-0 card, it creates a default STM profile for the card as well as 28 T1 profiles for each component T1 line.

The following example shows the parameters in a STM profile, with the example settings:

```
admin> read stm { 1 7 1 }
STM/{ shelf-1 slot-7 1 } read
  name = ""
  physical-address* = { shelf-1 slot-7 0 }
  loop-timing = yes
```

Parameter	Specifies
Name	A profile name of up to 16 characters. The name is displayed after the line's physical address in the Dir command output.
Physical-Address	Location of the card in the system.
Loop-Timing	Clock source for the line. By default, an STM-0 line uses loop-timing, which means the line derives its timing from the input clock. When loop-timing is set to No, the line derives its timing from the TAOS unit's internal clock. Lucent recommends that you use the default Loop-Timing setting.

For information about T1 line profiles, see Chapter 7, "Configuring T1 Cards."

Sample STM-0 configurations

To configure the STM-0 card, you must configure each component T1 profile. In most cases, you do not need to modify the default configuration of the STM-0 card.

Example of configuring an STM profile

Note: Use of the internal clock is generally not recommended.

To configure the lines of a STM-0 card to use its own internal clock for the timing of the line:

```
admin> read stm {1 1 1}
STM/{ shelf-1 slot-1 1 } read
admin> set loop-timing = no
admin> write
STM/{ shelf-1 slot-1 1 } written
```

After configuring the STM-0 line, configure the individual T1 lines that constitute the STM-0 line as explained in the next section.

Example of configuring a T1 data trunk

The following commands configure a T1 line as an SS7 data trunk, enabling the signaling gateway to control the line:

```
admin> read t1 {1 1 7}
T1/{ shelf-1 slot-1 7 } read

admin> set line-interface enabled = yes

admin> set line-interface signaling-mode = ss7-data-trunk

admin> set line-interface incoming-call-handling = internal-processing

admin> write
T1/{ shelf-1 slot-1 7 } written
```


Configuring DSL Connections (DSLNT)

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Introduction to DSL technologies

Three types of Digital Subscriber Line (DSL) cards are available for the DSLNT. These cards support either Asymmetric Digital Subscriber Line (ADSL), ISDN Digital Subscriber Line (IDSL) or Symmetric Digital Subscriber Line (SDSL). A summary of each protocol follows.

IDSL overview

IDSL is part of a broad range of MultiDSL™ offerings that let you implement DSL technologies immediately. Because IDSL uses the same 2B1Q signaling used by ISDN, existing ISDN U-interface devices—such as terminal adapters (TAs) or Lucent Pipelines® products—can connect to a DSLNT with an IDSL line card without modification.

IDSL supports high-bandwidth applications such as remote access, Internet or intranet access, and telecommuting. This integrated solution provides centralized line terminations to single-pair copper wires for transmission of full-duplex data at 128 Kbps and at distances of up to 18,000 feet (5.5km).

The IDSL DSLNT provides a separate network that does not congest the “voice network” with data traffic. In this way, the DSLNT replaces a switch for data traffic. The IDSL line card also provides some of the functionality of a switch for monitoring line quality and troubleshooting the line. As subscriber requirements change, you can use the same platform to add other MultiDSL technologies such as SDSL and ADSL.

By installing the IDSL line card into the DSLNT, you can cost-effectively support a wide range of analog, ISDN, Frame Relay and IDSL services on a single, manageable platform.

The IDSL line card provides the following features:

- ISDN BRI (2B1Q) signaling
- Two DB37 connectors, each providing 16 IDSL sessions for a total of 32 sessions
- 128Kbps user bit rate over a two-wire subscriber loop
- Line Termination (LT) mode
- No switch required
- Point-to-point connectivity
- Support for both switched channels and nailed channels (including Lucent's SuperDigital 128)
- Support for maintenance functions including BRI-U interface monitoring commands, loopback, and out-of-band management

IDSL supports many of the same configuration options as other types of connections, such as nailed and switched sessions, PPP, MP and MPP encapsulation, and incoming and outgoing voice calls.

ADSL overview

Asymmetric Digital Subscriber Line (ADSL) supports high-bandwidth applications such as remote access, Internet or intranet access, and telecommuting. The DSLTNT supports both Carrierless Amplitude Modulation (CAP) and Discrete Multitone (DMT) standards. Both standards support rate adaption, which enables the DSLTNT to detect the noise level on the line and automatically adjust the data transfer rate for optimum performance.

The DSLTNT ADSL cards also support the MultiDSL voice splitter. The voice splitter solution works in conjunction with Lucent DSLPipe™ products to integrate Plain Old Telephone Service (POTS) with ADSL data.

The ADSL-CAP card supports the following asymmetric transfer rates:

Upstream rate	Downstream rate	Distance
544 Kbps	640 Kbps	17,000 feet (5.18 km)
1.088 Mbps	2.560 Mbps	12,000 feet (3.66 km)
1.088 Mbps	7.168 Mbps	10,000 feet (3.05 km)

The ADSL-DMT card supports the following maximum asymmetric transfer rates:

Wire gauge (AWG)	Upstream rate	Downstream rate	Distance
24	704 Kbps	3040 Kbps	17,000 feet (5.18 km)
26	192 Kbps	512 Kbps	17,000 feet (5.18 km)
24	896 Kbps	7584 Kbps	12,000 feet (3.66 km)
26	640 Kbps	3904 Kbps	12,000 feet (3.66 km)

Wire gauge (AWG)	Upstream rate	Downstream rate	Distance
24	928 Kbps	9248 Kbps	10,000 feet (3.05 km)
26	736 Kbps	6976 Kbps	10,000 feet (3.05 km)

SDSL overview

The SDSL card supports symmetric data transfer rates of 768 Kbps for a distance of up to 12,000 feet (3.7 km) over a single pair of copper wires.

The SDSL-HS data card expands on features offered in the 16-port SDSL card. The SDSL-HS data card provides high port density, with 24 SDSL lines per card. The card supports high-speed symmetric data transfer, with rates up to 1.5 Mbps and distances to 14,000 feet (4.3km), through a single pair of copper wires. At reduced data transfer rates, the card supports distances over 18,000 feet (5.5km). The SDSL-HS data card has an SDSL chip and board layout that differ from the 16-port SDSL card and that provide high speed, multirate capabilities. The 16-port SDSL card cannot be upgraded to the SDSL-HS technology.

The SDSL-HS card is compatible with the DSLPipe products but must be set to 768 Kbps to work properly with the DSLPipe-S, or to 400, 784 or 1168 Kbps to work with the DSLPipe-2S.

SDSL supports Frame Relay and Point-to-Point Protocol (PPP). You configure Frame Relay or PPP connections on an SDSL connection in the same way you configure them on a T1 or serial WAN (SWAN) interface.

The SDSL-HS card supports the following symmetric transfer rates:

Wire gauge (AWG)	Data transfer rate	Distance
24	400 Kbps	18,000 feet (5.5 km)
26	400 Kbps	14,500 feet (4.4 km)
24	784 Kbps	18,000 feet (5.5 km)
26	784 Kbps	13,000 feet (3.96 km)
24	1.168 Mbps	16,000 feet (4.88 km)
26	1.168 Mbps	11,000 feet (3.35 km)
24	1.5 Mbps	13,000 feet (3.96 km)
26	1.5 Mbps	10,000 feet (3.05 km)

Note: The data transfer rates presented in the table above are approximations. Actual data transfer rates depend on line loop quality and can vary.

For complete details of configuring authentication or Frame Relay and PPP connections on your DSLNT, see the *APX 8000/MAX TNT/DSLNT WAN, Routing and Tunneling Configuration Guide* and the *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide*.

DSL configuration

You configure DSL connections in much the same way you configure ISDN or modem connections. DSL connections can be configured as switched or nailed PPP, MP, or MPP, or as Frame Relay-encapsulated connections. You can also use your existing authentication methods, such as RADIUS, to authenticate DSL users, by using PPP protocols in conjunction with PAP or CHAP. You can do this either when the units are first turned on or by setting an inactivity timer.

Any ISDN TA or router (such as a Lucent Pipeline) that supports ISDN BRI (2B1Q) signaling can be connected over an IDSL connection. ADSL and SDSL connections require Ascend DSLPipe units on the remote end.

DSL connections require the following general configuration on the DSLNT:

- The DSL port in the line profile
- A Connection profile for the remote device
- For Frame Relay connections, a Frame Relay profile

In addition to standard routing connections, you can configure the following DSL-specific capabilities:

- DSLPipe plug and play
- IDSL voice support

Note: For better system performance, Lucent recommends that you enable only DSL ports that are in use, (By default, DSL ports are disabled.)

Configuring switched connections

A DSL physical link is always up, but a PPP session can be established and terminated based on data activity, just as it is for ISDN or PSTN calls. Each PPP session initiates negotiations, followed by authentication and accounting. Switched connections can provide per session authentication as well as accounting information typically used for client billing.

From the service provider perspective, a DSL connection is handled exactly like an ISDN or PSTN call. The DSLNT checks the Answer-Defaults profile, applies authentication methods, and establishes the PPP session. After some inactivity, the PPP session is dropped, again generating accounting information. DSLPipe units initiate all switched ADSL and SDSL connections, and the DSLNT handles them as regular incoming PPP calls. Note that Frame Relay connections must be nailed.

You configure the DSLPipe for a switched connection in a similar way to other Pipeline switched connections, with the following important differences:

- Set the Chan Usage parameter in the Configure profile to Switch/Unused (for ADSL or SDSL connections) or Switch/Switch (for IDSL connections).

- Set the Dial # parameter in the Configure profile to the DSL port number, which in the case of a single DSLPipe is always 1.

To configure a switched connection on the DSLTNT for an incoming connection from a DSLPipe, you must set the Call-Type parameter to Off in the Connection profile for the DSLPipe. For example:

```
admin> read connection dslpipe-1
CONNECTION/dslpipe-1 read

admin> set telco call-type = off

admin> write
CONNECTION/dslpipe-1 read
```

Configuring nailed connections

In a nailed connection, the DSLTNT and the remote unit always assume that the connection is up and do not attempt to verify that the line is operational.

A nailed connection does not record accounting or authentication information after the session is established and therefore cannot be used to bill for DSL service as if it were a call on an ISDN network or the PSTN.

Nailed connections are typically used for Frame Relay connections, but PPP can also be used. Voice calls are not supported over a nailed connection.

You specify whether an ADSL or SDSL connection is nailed by doing the following:

- Specifying a nailed group number in the ADSL or SDSL profile
- Setting Call-Type to FT1 in the Connection profile for the nailed connection

You specify whether an IDSL connection is nailed by doing the followings:

- Specifying a nailed group number in the IDSL profile
- Setting Channel-Usage to Nailed-64-Channel in the IDSL profile
- Setting Call-Type to FT1 in the Connection profile for the nailed connection

You configure the DSLPipe for a nailed connection in a similar way to other Pipeline nailed connections:

- In the Configure profile, set Chan Usage to Leased/Unused.
- In the Connection profile for the DSLTNT, set Call Type to Nailed in the Telco Options submenu.
- In the Connection profile for the DSLTNT, specify a Group number in the Telco Options submenu.

“Sample DSL configurations” on page 17-22, provides examples of configuring nailed DSL connections.

Configuring data transfer rates

You can configure DSL upstream and downstream rates in the line profiles for each card, and in Connection or RADIUS profiles. The data transfer rates in the line profiles apply to the port. The data rate limits in Connection or RADIUS profiles apply only to sessions using that particular profile.

Configuring session rate limits enables you allocate portions of a DSL connection's bandwidth to particular users. For information, see "Configuring per-session data transfer rates" on page 17-8.

Table 17-1 describes the parameters that determine the data transfer rates on the DSLNTNT. For detailed information about these parameters, see the *APX 8000/MAX TNT/DSLNTNT Reference*.

Table 17-1.DSL data rate configuration parameters

Parameter	Cards it applies to
SDSL line profile Data-Rate-Mode Max-Rate	SDSL 24 port SDSL only
ADSL-CAP or ADSL-DMT line profile Data-Rate-Mode Max-Up-Stream-Rate Max-Down-Stream-Rate	ADSL-CAP, ADSL-DMT ADSL-CAP, ADSL-DMT ADSL-CAP, ADSL-DMT
Connection profile > Session-Options Ses-ADSL-Dmt-Up-Rate Ses-ADSL-Dmt-Down-Rate Ses-ADSL-Cap-Up-Rate Ses-ADSL-Cap-Down-Rate Ses-Rate-Mode Ses-Rate-Type Ses-SDSL-Rate Rx-Data-Rate-Limit Tx-Data-Rate-Limit	ADSL-DMT ADSL-DMT ADSL-CAP ADSL-CAP ADSL-CAP, ADSL-DMT, SDSL ADSL-CAP, ADSL-DMT, SDSL SDSL SDSL SDSL

Configuring data transfer rates for ADSL lines

The Max-Down-Stream-Rate parameter in the ADSL-DMT and ADSL-CAP line profiles specifies the maximum downstream rate that the transceiver supports. If loop quality is poor, the transceiver chooses the lower rates, and good loop quality causes the transceiver to choose the higher rates. If the loop quality is very poor, the transceiver will not train at all, and will be unable to connect to the remote side. In that case, you must specify a lower maximum downstream rate, because the transceiver does not cross rate boundaries.

For example, if the transceiver is configured for 1088000bps and the loop quality is so poor that the transceiver will not connect to the remote side, the transceiver does not automatically adjust the down-rate into the 952000bps range. The administrator needs to configure the Max-Down-Stream-Rate to the lower rate.

Note: For more information about the Max-Downstream-Rate parameter, see the *APX 8000/MAX TNT/DSLNT Reference*. Note that although the Max-Upstream-Rate parameter appears in the ADSL-CAP and ADSL-DMT profiles, it is not currently supported.

To configure the maximum data rate for an ADSL connection, proceed as in the following example:

- 1 Read in the ADSL-CAP or ADSL-DMT profile:

```
admin> read adsl-cap {2 3 2}
ADSL-CAP/{ shelf-2 slot-3 2 } read
```
- 2 Enable the line:

```
admin> set enabled=yes
```
- 3 List the Line-Config profile:

```
admin> list line-config
[in ADSL-CAP/{ shelf-2 slot-3 2 }:line-config]
trunk-group = 0
nailed-group = 1
activation = static
call-route-info = { any-shelf any-slot 0 }
data-rate-mode = autobaud
max-up-stream-rate = 1088000
max-down-stream-rate = 2560000
```
- 4 Specify a maximum downstream rate:

```
admin> set max-down-stream-rate=5120000
```
- 5 Write the profile:

```
admin> write
```

Configuring data transfer rates for SDSL lines

The 16-port SDSL card only supports a maximum symmetric data transfer rate of 784Kbps. You can, however, configure the 24-port SDSL-HP card maximum data rate using the Max-Rate parameter in the SDSL line profile. The Max-Rate parameter supports the following values:

- 144000
- 272000
- 400000
- 528000
- 784000
- 1168000
- 1552000

Configuring DSL Connections (DSLNT)

Configuring data transfer rates

To configure the maximum transfer rate for the 24-port SDSL card:

- 1 Open the SDSL profile:

```
admin>read sdsl {2 1 7}
SDSL/{ shelf-2 slot-1 7 } read
```

- 2 Enable the line:

```
admin>set enabled=yes
```

- 3 List the Line-Config profile:

```
admin> list line-config
[in SDSL/{ shelf-2 slot-1 7 }:line-config]
trunk-group = 0
nailed-group = 1
activation = static
call-route-info = { any-shelf any-slot 0 }
data-rate-mode = singlebaud
max-rate = 784000
unit-type = coe
```

- 4 Specify a maximum rate:

```
admin> set max-rate=1552000
```

- 5 Write the profile:

```
admin> write
```

Configuring per-session data transfer rates

The DSL cards support configuring per-session data transfer rates for individual DSLPipe customer premises equipment (CPE) user sessions.

You can use two different methods to configure the per-session data transfer rates for DSL connections: modem rate control and data-rate limits.

In modem rate control, the DSLTNT initially establishes a CPE session at the maximum available data rate. If the CPE specifies a lower data rate, the DSLTNT terminates the session, then reestablishes it at the rate specified by the CPE. The next time the CPE initiates a connection, the DSLTNT does not retrain if the initial rate is the same or lower than the rate used previously for that CPE.

In data-rate limit, you specify transmit and receive data rate limits that apply to logical sessions on the DSL line. Data-rate limits enable multiple individual sessions on each DSL line.

Following are the Connection profile parameters for configuring per-session data rates:

Parameter/RADIUS attribute	Specifies
Ses-Rate-Type/ Ascend-Dsl-Rate-Type (92)	Type of DSL connection to rate control. Disabled (the default) means that modem rate control is not active for the connection. Currently, Disabled and ADSL-CAP settings are the only supported options.

Parameter/RADIUS attribute	Specifies
Ses-Rate-Mode/ Ascend-Dsl-Rate-Mode (97)	Per-session DSL data rate mode. The default setting, Autobaud, specifies that the DSLTNT must train up to a set data rate. If a DSL modem cannot train to this data rate, it will connect at the closest rate to which it can train (the modem's ceiling rate). Currently Autobaud is the only supported option.
Ses-ADSL-CAP-Up-Rate/ Ascend-DSL-Upstream-Limit (98)	Per-session ADSL-CAP upstream data rate. <i>Not currently supported.</i>
Ses-ADSL-CAP-Down-Rate/ Ascend-DSL-Downstream-Limit (99)	Per-session ADSL-CAP downstream data rate. The following rates (in bits per second) are supported: 7168000 (the default), 6272000, 5120000, 4480000, 3200000, 2688000, 2560000, 2240000, 1920000, 1600000, 1280000, 960000, 640000.
Ses-ADSL-DMT-Up-Rate/ N/A	Not currently supported.
Ses-ADSL-DMT-Down-Rate/ N/A	Not currently supported.
Rx-Data-Rate-Limit/ N/A	Maximum data rate (in kilobits per second) to be received across the connection. The default 0 (zero) disables the data rate limit feature. The valid range is from 0 to 64000. If the specified number is larger than the actual bandwidth provided by the line, the connection behaves as if the data rate limit were disabled, except that additional computations are performed unnecessarily.
Tx-Data-Rate-Limit/ N/A	Maximum data rate in kilobits per second to be transmitted across the connection. The default 0 (zero) disables the data rate limit feature. The valid range is from 0 to 64000. If the specified number is larger than the actual bandwidth provided by the line, the connection behaves as if the data rate limit were disabled, except that additional computations are performed unnecessarily.

For more information about these parameters, see the *APX 8000/MAX TNT/DSLNT Reference*.

Configuring per-session data rates using modem rate control

In the following example, the CPE session will be initially established at the maximum line rate configured in the ADSL-CAP profile. After the session has been established, the DSLTNT determines that this session has a maximum downstream rate of 7168000. It then re-establishes the connection using the specified rate.

```
admin> read conn adslpipe-1
CONNECTION/adslpipe-1 read
```

Configuring DSL Connections (DSLNT)

Configuring data transfer rates

```
admin> set session ses-rate-type = adsl-cap
admin> set session ses-adsl-cap-down-rate = 7168000
admin> write
CONNECTION/adslpipe-1 written
```

Following is a comparable RADIUS profile:

```
adslpipe-1 Password = "pipepw", User-Service = Framed-User
Framed-Protocol = PPP,
Framed-Address = 10.2.3.31,
Framed-Netmask = 255.255.255.0,
Ascend-Dsl-Rate-Type = Rate-Type-AdslCap,
Ascend-Dsl-Rate-Mode = Rate-Mode-AutoBaud,
Ascend-Dsl-Downstream-Limit = adslcap-dn-7168000
```

Configuring per-session data rate limits

You can configure transmit and receive data rate limits for individual connections that use the CAP-ADSL, SDSL, and unchannelized DS3 cards. ISPs can use these configuration parameters to limit bandwidth for a connection according to the rate charged for the account.

Note: If the parameters are set for a connection that does not use these cards, the system ignores the settings.

To configure an SDSL per-session data rate, proceed as in the following example:

- 1 Read in a Connection profile

```
admin> read connection sdsl-1
CONNECTION/sdsl-1 read
```
- 2 List the Session-Options profile:

```
admin> list session-options
[ in CONNECTION/sdsl-1:session-options ]
..
..
..
rx-data-rate-limit = 0
tx-data-rate-limit = 0
```
- 3 Specify a maximum receive rate:

```
admin> set rx-data-rate-limit=64000
```
- 4 Specify a maximum transmit rate:

```
admin> set tx-data-rate-limit=64000
```
- 5 Write the profile:

```
admin> write
```

Sample log session showing rate control negotiation

The following log messages show an incoming call from the user named adslpipe-1. The connection is authenticated via RADIUS. After establishing the LAN session, the DSLTNT reads the data rates:

LOG info, Shelf 1, Controller, Time: 16:47:11--
[1/7/1/1] Incoming Call [MBID 18]

LOG info, Shelf 1, Controller, Time: 16:47:11--
[1/7/1/0] Assigned to port [MBID 18]

LOG info, Shelf 1, Controller, Time: 16:47:11--
[1/7/1/1] Call Connected [MBID 18]

LOG info, Shelf 1, Slot 7, Time: 16:47:14--
[1/7/1/0] LAN session up: <adslpipe-1> [MBID 18]
[adslpipe-1]

LOG notice, Shelf 1, Slot 7, Time: 16:47:14--
Line 1 (radius) profile adslpipe-1
from <autobaud,1088000,2560000>
to <autobaud,952000,7168000>

LOG notice, Shelf 1, Slot 7, Time: 16:47:14--
Line 1 (radius) profile adslpipe-1 operation rates
<autobaud,1088000,2560000>

The DSLNT then terminates the call and re-establishes it using the configured data rates:

LOG notice, Shelf 1, Slot 7, Time: 16:47:14--
Reconfigure Line 1 (radius) profile adslpipe-1

LOG notice, Shelf 1, Slot 7, Time: 16:47:14--
Line 1 OOS

LOG warning, Shelf 1, Controller, Time: 16:47:14--
[1/7/1/1] Call Disconnected [MBID 18]

LOG info, Shelf 1, Controller, Time: 16:47:14--
[1/7/1/0] Call Terminated [MBID 18]

LOG notice, Shelf 1, Slot 7, Time: 16:47:14--
Line 1 INS

LOG info, Shelf 1, Slot 7, Time: 16:47:14--
[1/7/1/0] LAN session down: <adslpipe-1> [MBID 18]
[adslpipe-1]

LOG warning, Shelf 1, Slot 7, Time: 16:47:14--
[1/7/1/0] STOP: 'adslpipe-1'; cause 185.; progress 60.;
host 200.200.200.1 [MBID 18] [adslpipe-1]

LOG notice, Shelf 1, Slot 7, Time: 16:47:30--
Line 1 up

LOG info, Shelf 1, Controller, Time: 16:47:34--
[1/7/1/1] Incoming Call [MBID 19]

```
LOG info, Shelf 1, Controller, Time: 16:47:34--  
[1/7/1/0] Assigned to port [MBID 19]  
  
LOG info, Shelf 1, Controller, Time: 16:47:34--  
[1/7/1/1] Call Connected [MBID 19]  
  
LOG info, Shelf 1, Slot 7, Time: 16:47:39--  
[1/7/1/0] LAN session up: <adslpipe-1> [MBID 19]  
[adslpipe-1]  
  
LOG notice, Shelf 1, Slot 7, Time: 16:47:39--  
Line 1 (radius) profile adslpipe-1  
from <autobaud,1088000,2560000>  
to <autobaud,952000,7168000>  
  
LOG notice, Shelf 1, Slot 7, Time: 16:47:39--  
Line 1 (radius) profile adslpipe-1 successfully retrained  
<autobaud,952000,7168000>
```

Configuring DSLPipe Plug and Play

The Plug and Play feature enables a DSLPipe to obtain its configuration through the DSLTNT by using the Dynamic Host Configuration Protocol (DHCP) and Trivial File Transfer Protocol (TFTP). The DSLPipe ships with the Plug and Play feature enabled, so it requires no configuration if the DSLTNT and servers have been configured properly.

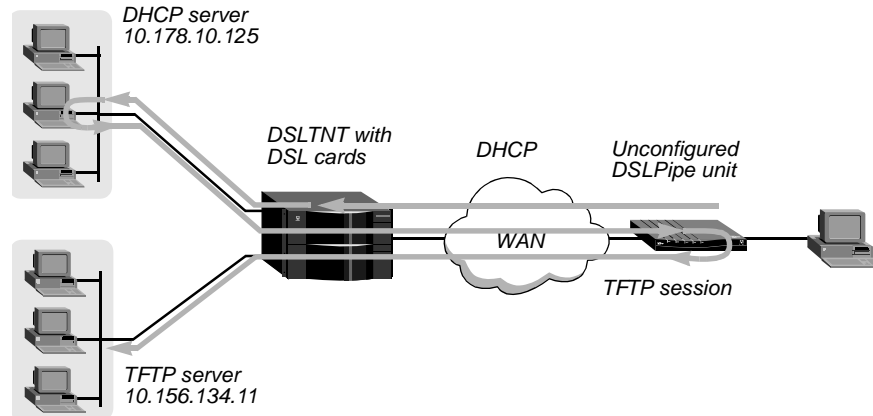
How Plug and Play works

When the DSLPipe unit initially comes up, it uses factory default settings that enable it to forward a DHCP request to a DSLNT, which sends the request to a DHCP server. The connection between the DSLTNT and the DSLPipe is a nailed Frame Relay-encapsulated connection configured for bridging.

The DHCP server returns an IP address, netmask, the path to a more detailed configuration file, and a TFTP server hostname. The DSLTNT forwards the DHCP response to the requesting DHCP client.

Figure 17-1 illustrates the Plug and Play feature.

Figure 17-1. DSLPipe unit obtaining its configuration (Plug and Play)



The DSLPipe unit uses the minimal configuration it obtains via DHCP to access the specified TFTP server and a configuration file, which is identified by a filename that matches the unit's serial number. After downloading the file, the DSLPipe begins using the configuration.

For this feature to work, the network administrator must set up the DHCP and TFTP servers, as described in the next sections. In addition, the DHCP server must be configured to access DNS, so the client can access the specified TFTP server by name.

For details about the DSLTNT requirements, see "Configuring the DSLTNT" on page 17-15.

DHCP server requirements

The following sample configuration shows required DHCP settings for a Pipeline 130 unit acting as a DHCP server. Other DHCP server implementations might have additional requirements. This example shows only the DHCP-related settings in the Ethernet Mod Config profile:

```
20-B00 Mod Config
DHCP Spoofing...
  DHCP Spoofing=Yes
  DHCP PNP Enabled=Yes
  Renewal Time=10
  Become Def. Router=No
  Dial if Link Down=No
  Always Spoof=Yes
  Validate IP=No
  Maximum no reply wait=10
  IP Group 1=10.10.10.1/16
  Group 1 count=10
  IP group 2=0.0.0.0/0
  Group 2 count=0
  Host 1 IP=0.0.0.0/0
  Host 1 Enet=000000000000
  Host 2 IP=0.0.0.0/0
  Host 2 Enet=000000000000
  Host 3 IP=0.0.0.0/0
  Host 3 Enet=000000000000
```

```
TFTP Host Name=host-1.abc.com  
Boot File Path=/tftpboot/config
```

For a Pipeline unit to operate as a DHCP server, DHCP Spoofing and Always Spoof must be set to Yes. To enable the server to return an IP address, netmask, path to a more detailed configuration file, and TFTP server name, configure the following parameters:

- Set the IP Group 1 parameter and Group 1 Count parameters to represent a valid IP address pool.
- Set the TFTP Host Name parameter to the hostname of the TFTP server on which the DSLPipe configurations reside.
- Set the Boot File Path parameter to the directory on the TFTP server that contains the DSLPipe configurations.

For details on the other settings, see the documentation for the Pipeline unit.

TFTP server requirements

In this sample configuration, the TFTP server uses the /tftpboot/config directory to store configuration files. This is consistent with the DHCP configuration shown in the preceding section, which passes the following pathname to the DSLPipe client:

```
/tftpboot/config
```

The filename of a DSLPipe configuration file must match the unit's serial number. DSLPipe serial numbers are located on a label on the bottom of the unit and in the 00-100 status window.

DSLPipe default configuration

In its default configuration, the DSLPipe is configured as follows:

In this menu:	These are the defaults:
Configure	Route=None Bridge=Yes My Name=DSLPipe
Ethernet > Connections	Station=DSLPipe Active=Yes Encaps=FR
Ethernet > Connections > Encaps Options	FR Prof = DSLframe DLCI=16
Ethernet > Frame Relay	Name=DSLframe Active=Yes FR Type=DTE Link Mgmt=T1.617D

Configuring the DSLNT

DSLPipe Plug and Play support requires the following configuration on the DSLNT:

- BOOTP Relay enabled
- A nailed DSL connection to the DSLPipe
- A Frame Relay profile that makes use of the DSL line
- A Connection profile for each DSLPipe unit

This section does not include the DSLNT IP and DNS configurations, which are required for Plug and Play to work. For details about configuring IP routing and DNS, see the *APX 8000/MAX TNT/DSLNT WAN, Routing and Tunneling Configuration Guide*.

Configuring BOOTP Relay

The DSLNT must be set up for BOOTP Relay to support Plug and Play in DSLPipe units. When you enable BOOTP Relay, the DSLNT can forward DHCP request packets to a DHCP server and forward DHCP responses back to the requesting client.

If more you specify more than one DHCP server, the DSLNT uses the first server until it becomes unavailable. Once it starts using the second DHCP server, it continues using that server until it becomes unavailable, at which time it switches back to using the first server again.

To enable BOOTP Relay, proceed as in the following example:

- 1 Read the IP-Global profile:

```
admin> read ip-global
IP-GLOBAL read
```
- 2 List the BOOTP-Relay profile:

```
admin> list bootp-relay
[in IP-GLOBAL:bootp-relay]
active = no
bootp-servers = [ 0.0.0.0 0.0.0.0 ]
```
- 3 Activate BOOTP Relay:

```
admin> set active=yes
```
- 4 Specify a DHCP server using the BOOTP-Servers setting. For example:

```
admin> set bootp-servers 1 =192.168.7.62
```
- 5 If necessary, specify a second DHCP server. For example:

```
admin> set bootp-servers 2 =192.168.7.72
```
- 6 Write the IP-Global profile to save your changes:

```
admin> write
IP-GLOBAL written
```

Configuring the SDSL profile

In the following example procedure, the network administrator configures an SDSL line in slot 3 of the DSLNT:

Configuring DSL Connections (DSLNT)

Configuring DSLPipe Plug and Play

- 1 Read in the SDSL profile. For example, if the SDSL card is installed in shelf 1, slot 11, and the remote DSLPipe is connected to port 1:

```
admin> read sdsl {1 3 1}
SDSL/{ shelf-1 slot-3 1 } read
```

- 2 List the profile:

```
admin> list
[in SDSL/{ shelf-1 slot-3 1 }]
name = ""
physical-address* = { any-shelf any-slot 0 }
enabled = no
line-config = { 0 1 static { any-shelf any-slot 0 } 144000 coe }
```

- 3 Enable the port:

```
admin> set enabled=yes
```

- 4 Assign this port to a nailed group:

```
admin> set line-config nailed-group=101
```

The Frame Relay profile you create next locates this port by the nailed group number. The nailed group must be unique for each active WAN interface.

- 5 Write the profile:

```
admin> write
SDSL/{ shelf-1 slot-3 1 } written
```

Configuring a Frame Relay profile

In the following example, the administrator creates a Frame Relay profile to be used by the Connection profile to connect to the DSLPipe:

See the *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide* for more details and examples.

To configure the Frame Relay profile:

- 1 Create a new Frame Relay profile:

```
admin> new frame-relay fr
```

- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Assign the Frame Relay profile to a nailed-up group:

```
admin> set nailed-up-group=101
```

This must be the same as the SDSL nailed group number you configured in the SDSL profile. The nailed group must be unique for each active WAN interface.

- 4 Specify the type of link management used for the connection:

```
admin> set link-mgmt = ansi-t1.617d
```

This is the default for the DSLPipe.

- 5 Specify the type of link:

```
admin> set link-type = dce
```


- 6 Write the profile:

```
admin> write
FRAME-RELAY/fr written
```

Configuring a Connection profile

This example Connection profile uses the Frame Relay profile configured in the previous section to reach the DSLPipe.

- 1 Create a new Connection profile:

```
admin> new connection dslpipe
CONNECTION/dsl-pipe read
```

- 2 Activate the profile:

```
admin> set active = yes
```

- 3 Specify Frame Relay as the encapsulation used on the link:

```
admin> set encapsulation-protocol = frame-relay
```

- 4 Specify the IP address that will be assigned to the DSLPipe unit:

```
admin> set ip-options remote-address = 11.10.10.1/16
```

- 5 Specify that only nailed channels are used on this link:

```
admin> set telco-options call-type = ft1
```

- 6 Specify the name of the Frame Relay profile the Connection profile must use:

```
admin> set fr-options frame-relay-profile = fr
```

- 7 Specify the Frame Relay DLCI used for the connection:

```
admin> set fr-options dlci = 16
```

This is the DSLPipe unit's DLCI and the DSLTNT default.

- 8 Write the profile:

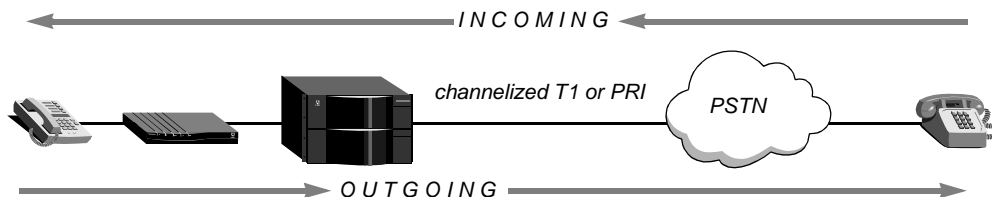
```
admin> write
CONNECTION/dslpipe read
```

Configuring IDSL voice connections

In this example (see Figure 17-2), a Pipeline connected via an IDSL line is configured for incoming and outgoing voice calls. The connection is a switched 128Kbps MP+ connection that allows the Pipeline to drop a data channel when it receives an incoming voice call, and bring the second data channel up again when the voice call is over. Voice calls are not supported over nailed connections.

This example uses a Lucent Pipeline, but you can configure any ISDN U-interface device, such as a terminal adapter (TA), similarly.

Figure 17-2. Incoming and outgoing voice calls



Incoming calls

Incoming IDSL voice calls require that the Central Office (CO) switch support Dialed Number Identification Service (DNIS). DNIS allows the DSLTNT to route incoming calls to the Pipeline or ISDN TA. The DSLTNT does this by comparing the DNIS number it receives to Answer-Number settings in IDSL profiles. When the Pipeline receives this incoming call, it routes the call to a particular phone on the basis of its own port assignments.

Outgoing calls

Outgoing calls require that you configure the DSLTNT to use trunk groups and that the ISDN TA or remote router (such as a Pipeline 75) support en-bloc sending.

Trunk groups assign DSLTNT T1 or E1 channels to groups that are identified by a number. When a user on the IDSL line prefaces the telephone number dialed with the trunk group number, the DSLTNT sends the call out on a channel in the trunk group.

With en-bloc sending, the Setup message that the DSLTNT forwards to the PSTN switch contains all information required to process the call, including the dialed number.

When a user dials out from an analog device connected to the analog port of the Pipeline or an ISDN TA, the user must use the trunk group number as the first digit of the telephone number. (This method is similar to dialing from locations where you must dial with an initial digit to get an outside line before entering the phone number.)

In addition, the user must terminate the telephone number with the pound (#) key. The Pipeline then sends a Q.931 En-Bloc Setup packet to the DSLTNT. The DSLTNT forwards the Setup message to the PSTN switch, which sets up the call.

Configuring the DSLTNT

To configure the DSLTNT for incoming and outgoing voice calls, you must configure the following:

- An IDSL profile
- A Connection profile for the remote device
- Trunk groups so the DSLTNT can send outgoing calls to the PSTN

Configuring the IDSL profile

To configure the DSLTNT IDSL profile, proceed as in the following example:

- 1 Read the IDSL profile the remote user is connected to. For example:

```
admin> read idsl {1 7 29}
IDSL/{ shelf-1 slot-7 29 } read
```

- 2 List the IDSL profile:

```
admin> list
name = 1:7:29
line-interface = { no [ { switched-channel 1 } { switched-channel +
physical-address* = { shelf-1 slot-7 29 } }
```

- 3 List the Line-Interface profile:

```
admin> list line-interface
[in IDSL/{ shelf-1 slot-7 29 }:line-interface]
enabled = no
channel-config = [ { switched-channel 1 } { switched-channel 1 } ]
answer-number-1 = ""
answer-number-2 = ""
clock-source = eligible
```

- 4 Enable the line:

```
admin> set enabled = yes
```

- 5 Specify the unique portion of the telephone number for the analog device attached to the Pipeline. The DSLTNT routes all calls it receives with this number to the device. For example, if a phone connected to a Pipeline unit has the number 510-555-1234, set the Answer-Number-1 parameter to the following value:

```
admin> set answer-number-1=5105551234
```

- 6 If two analog devices are attached to the Pipeline, configure the second IDSL channel with the appropriate phone number. For example:

```
admin> set answer-number-2=5105551235
```

- 7 Write the profile to save your changes:

```
admin> write
IDSL/{ shelf-1 slot-7 29 } written
```

Configuring a Connection profile for the remote device

To configure a Connection profile for the Pipeline:

- 1 Create a new Connection profile for the Pipeline:

```
admin> new connection pipeline
```

- 2 Activate the profile:

```
admin> set active=yes
```

- 3 Set the encapsulation to MP+ to allow the Pipeline to drop a data channel when it receives a voice call:

```
admin> set encapsulation-protocol=mpp
```

- 4 List the IP-Options profile:

```
admin> list ip-options
[in CONNECTION/pipeline:ip-options (new)]
ip-routing-enabled = yes
vj-header-prediction = yes
```

Configuring DSL Connections (DSLNT)

Configuring IDSL voice connections

```
remote-address = 0.0.0.0/0
local-address = 0.0.0.0/0
..
..
```

- 5 Specify the Pipeline IP address:

```
admin> set remote-address=192.1.2.1/24
```

- 6 List the Telco-Options profile:

```
admin> list.. telco-options
[in CONNECTION/pipeline:telco-options (new)]
answer-originate = ans-and-orig
callback = no
call-type = off
nailed-groups = 1
ftl-caller = no
force-56kbps = no
data-service = 56k-clear
..
..
```

- 7 Specify that the connection does not use nailed channels:

```
admin> set call-type=off
```

- 8 Set the data service:

```
admin> set data-service=64K-clear
```

- 9 Write the Connection profile:

```
admin> write
```

Configuring trunk groups

To enable the DSLTNT to recognize outgoing voice traffic and route it appropriately, you must use trunk groups. Note that when you enable trunk groups, you must configure every channel on the DSLTNT that will be used for outgoing calls with a trunk group.

To create trunk groups on the DSLTNT for IDSL outgoing calls:

- 1 Read the System profile:

```
admin> read system
SYSTEM read
```

- 2 Enable trunk groups:

```
admin> set use-trunk-groups = yes
```

- 3 Write the profile:

```
admin> write
SYSTEM written
```

- 4 Next, assign trunk groups to the lines used for placing outgoing calls. For example, to use T1 lines for outgoing calls, first read in the T1 profile:

```
admin> read t1 {1 1 1}
T1/{ shelf-1 slot-1 1 } read
```

- 5 List the Channel-Config subprofile :

```
techpubs-lab-25> list line-interface channel-config
[in T1/{ shelf-1 slot-1 1 }:line-interface:channel-config]
channel-config[1] = { switched-channel 9 "" { any-shelf any-slot +
channel-config[2] = { switched-channel 9 "" { any-shelf any-slot +
channel-config[3] = { switched-channel 9 "" { any-shelf any-slot +
channel-config[4] = { switched-channel 9 "" { any-shelf any-slot +
channel-config[5] = { switched-channel 9 "" { any-shelf any-slot +
..
..
```

- 6 Assign each T1 channel to a trunk group. For example:

```
admin> set 1 trunk = 5
admin> set 2 trunk = 5
admin> set 3 trunk = 5
admin> set 4 trunk = 5
admin> set 5 trunk = 5
admin> set 6 trunk = 5
admin> set 7 trunk = 5
admin> set 8 trunk = 5
admin> set 9 trunk = 5
admin> set 10 trunk = 5
..
..
```

This trunk group number must be prepended to the number dialed by users dialing out from the Pipeline.

- 7 Write the T1 profile:

```
admin> write
T1/{ shelf-1 slot-1 1 } written
```

Configuring the Pipeline

When configuring a remote ISDN device to attach to the IDSL line card, always select ATT 5ESS Point-to-Point as the switch type. The IDSL line card can only emulate the ATT 5ESS Point-to-Point switch. (If you are connecting using a Pipeline, you can specify an IDSL switch type. This selection emulates an ATT 5ESS Point-to-Point switch with en-bloc dialing support, which is required for IDSL voice calls.)

Before you configure the Pipeline, make sure the PC connected to the Pipeline has an IP address on the same subnet as the Pipeline, and that the IP address of the Pipeline is configured as the default gateway for the PC.

Configuring the Configure profile

The Pipeline Configure profile allows you to set up the basic parameters for a connection. To configure the Pipeline Configure profile:

- 1 From the Main Edit menu, select Configure.
- 2 Specify the following values:
 - Switch Type=**IDSL**
 - Chan Usage=**Switch/Switch**
 - My Num A=**55105554444**

- My Name=**buffer**
 - My Addr=**192.1.2.1/24**
 - Rem Name=**bufferstnt**
 - Rem Addr=**192.1.1.1/24**
 - Route=**IP**
- 3 Exit and save the Configure profile.
 - 4 Open Ethernet > Connections > **bufferstnt**
 - 5 Set the encapsulation to MP+:

Encaps=**MPP**
MP+ enables the Pipeline to drop one channel of a data call to answer the voice call instead of sending a busy signal. See the Pipeline documentation for details.
 - 6 Exit and save the Connection profile.

Sample DSL configurations

This section provides the following example DSL configurations:

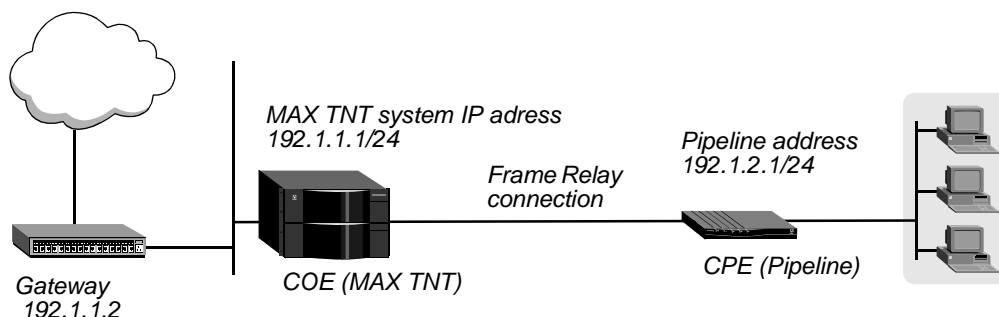
- An IDSL Frame Relay connection
- An ADSL nailed PPP connection
- An SDSL Frame Relay configuration using interface-based routing
- An SDSL Frame Relay configuration using system-based routing

Sample Frame Relay IDSL configuration

As Figure 17-3 illustrates, a Pipeline connects a single user to a DSLTNT over a 128Kbps nailed Frame Relay connection. It uses system-based routing. This example uses a Pipeline, but you can configure any ISDN U-interface device, such as a terminal adapter (TA), similarly. You must also assign that channel a group number using the Nailed-Group parameter. The Connection profile for the remote device then refers to the assigned group number in its Nailed-Group setting to direct the connection to use the IDSL nailed channel.

Note: This configuration does not support voice calls. For information on configuring an IDSL connection to support voice, see “Configuring IDSL voice connections” on page 17-17.

Figure 17-3. IDSL connection with a Pipeline



Configuring the DSL TNT

This example assumes the DSL TNT has already been configured with the following information:

- IP address of 192.1.1 4/24
- System name of `idslntnt`

To configure the DSL TNT for this example you must configure the following:

- A Connection profile for the remote device
- An IDSL profile
- A Frame Relay profile
- A static route to the gateway

Configuring a Connection profile for the remote device

To configure a Connection profile for the remote device:

- 1 Create a Connection profile for the Pipeline:
`admin> new connection pipeline`
- 2 Activate the profile:
`admin> set active=yes`
- 3 Set the encapsulation:
`admin> set encapsulation-protocol=frame-relay`
- 4 List the IP-Options profile:
`admin> list ip-options`
- 5 Enable IP routing for this Connection profile:
`admin> set ip-routing-enabled=yes`
- 6 Specify the Pipeline IP address:
`admin> set remote-address=192.1.2.1/24`
- 7 List the FR-Options profile:
`admin> list .. fr-options`
- 8 Specify the name of the Frame Relay profile:
`admin> set frame-relay-profile=idslntnt-fr`
- 9 Specify the Frame Relay DLCI:
`admin> set dlci=16`
- 10 List the Telco options profile:
`admin> list .. telco-options`
- 11 Set the data service:
`admin> set data-service=64K-clear`
- 12 Specify that the connection uses nailed channels:
`admin> set call-type=ft1`
- 13 Write the Connection profile:

```
admin> write
```

Configuring the IDSL profile

To configure the DSLTNT IDSL profile, proceed as in the following example:

- 1 Read the IDSL profile that the remote user is connected to. For example:

```
admin> read idsl {1 7 18}
IDSL/{ shelf-1 slot-7 18 } read
```

- 2 Enable the line:

```
admin> set line enabled = yes
```

- 3 List the configuration for the first channel:

```
admin> list line channel 1
[in IDSL/{ shelf-1 slot-7 18 }:line-interface:channel-con +
channel-usage = switched-channel
nailed-group = 0
```

- 4 Specify that the connection is nailed:

```
admin> set channel-usage = nailed-64-channel
```

- 5 Specify the nailed group. This group is referred to in the Connection profile for the remote device so the DSLTNT can determine which interface to use for the connection:

```
admin> set nailed-group = 10
```

- 6 Configure the second channel as nailed and assign it the same group number. For example:

```
admin> list .. 2
[in IDSL/{ shelf-1 slot-7 18 }:line-interface:channel-con +
channel-usage = switched-channel
nailed-group = 0
```

```
admin> set channel-usage = nailed-64-channel
```

```
admin> set nailed-group = 10
```

- 7 Write the profile to save your changes:

```
admin> write
IDSL/{ shelf-1 slot-7 18 } written
```

Configuring the Frame Relay profile

To configure the Frame Relay profile:

- 1 Create a new Frame Relay profile:

```
admin> new frame-relay idsltnt-fr
```

- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Assign the Frame Relay profile to a nailed-up group:

```
admin> set line nailed-up-group=10
```

This value must be the same as the IDSL nailed group number you configured in the IDSL profile. The nailed group must be unique for each active WAN interface.

- 4 Write the profile:


```
admin> write
```

Configuring a static route to the gateway

To configure a static route to the gateway:

- 1 Read in the IP-Route Default profile:

```
admin> read ip-route default
```
- 2 Enter the address of the Gateway on the local LAN to the remote network.

```
set gateway-address = 192.1.1.2
```
- 3 Write the profile:

```
admin> write
```

Configuring the Pipeline

Note: When configuring a remote ISDN device to attach to the IDSL line card, always select ATT 5ESS Point-to-Point as the switch type. The IDSL line card can only emulate the ATT 5ESS Point-to-Point switch. (On a Pipeline, you can specify an IDSL switch type. This selection emulates an ATT 5ESS Point-to-Point switch with en-bloc dialing support, which can be used for IDSL voice calls.)

Before you configure the Pipeline, make sure the PC connected to the Pipeline has an IP address on the same subnet as the Pipeline, and that the IP address of the Pipeline is configured as the default gateway for the PC.

Configuring the Configure profile

The Pipeline Configure profile allows you to set up the basic parameters for a connection. To configure the Pipeline Configure profile:

- 1 From the Main Edit menu, select Configure.
- 2 Specify the following values:
 - Switch Type=**IDSL**
 - Chan Usage=**Leased/Unused**
 - My Name=**pipeline**
 - My Addr=**192.1.2.1/24**
 - Rem Name=**idsltnt**
 - Rem Addr=**192.1.1.1/24**
 - Route=**IP**
- 3 Exit and save the Configure profile.

Configuring the Frame Relay profile

The Frame Relay profile defines the physical link used by the Connection profile to connect to the DSL TNT. To configure the Frame Relay profile:

- 1 Open the Ethernet > Frame Relay > *any profile*
- 2 Specify the following values:

Configuring DSL Connections (DSLNTNT)

Sample DSL configurations

- Name=**idsl-fr**
- Active=**Yes**
- Call Type=**Nailed**
- Nailed Grp=**1**

3 Exit the Frame Relay profile and save your changes.

Note that the Pipeline uses the following nailed group numbers:

- 1 is the first B channel
- 2 is the second B channel

Configuring the Connection profile

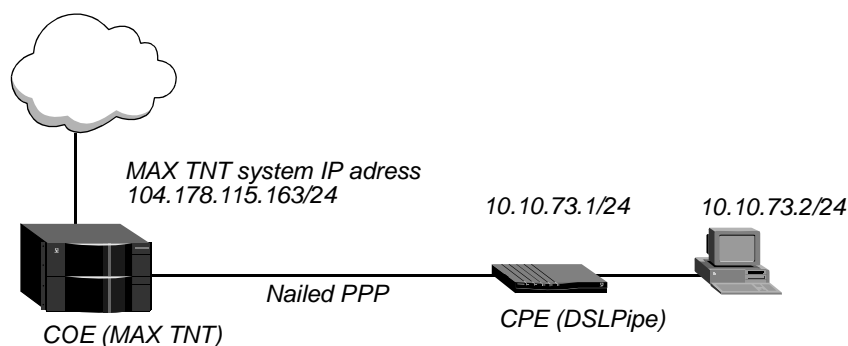
You must configure other, specialized options in the Connection profile for the DSLTNT, including the name of the Frame Relay profile and the nailed group assigned to it. To do this, proceed as in the following example:

- 1 Open Ethernet > Connections > **idslntnt**
- 2 Specify Frame Relay encapsulation:
Encaps=**FR**
- 3 Open the Encaps Options submenu.
- 4 Specify name of the Frame Relay profile used by this connection and a DLCI.
 - FR Prof=**idsl-fr**
 - DLCI=**16**
- 5 Exit and save the Connection profile.

Sample ADSL nailed PPP connection

In Figure 17-4, a DSLPipe connects to a DSLTNT ADSL card over a nailed PPP connection. The ADSL card is in slot 7, and the DSLPipe is connected to port 3 of the ADSL card. The DSLPipe IP address is 10.10.73.1/24. The DSLTNT IP address is 104.178.115.163/24. This example uses ADSL, but you can configure an SDSL connection similarly.

Figure 17-4. Sample ADSL PPP connection



Configuring the ADSL profile

To configure the ADSL profile in this example:

- 1 Read in the ADSL profile:

```
admin> read adsl-cap {1 7 3}
```

- 2 Enable the port:

```
admin> set enabled=yes
```

- 3 List the contents of the Line-Config profile:

```
admin> list line-config

[in ADSL-CAP/{ shelf-1 slot-7 3 }:line-config]
trunk-group = 0
nailed-group = 0
activation = static
call-route-info = { any-shelf any-slot 0 }
max-down-stream-rate = 7168000
```

- 4 Assign this port to a nailed group:

```
admin> set nailed-group=73
```

This nailed group points to the Connection profile you will create later. The nailed group must be unique for each active WAN interface.

- 5 Specify the maximum downstream rate:

```
admin> set max-down-stream-rate=7168000
```

- 6 Write the profile:

```
admin> write
```

Configuring the Connection profile

To configure the Connection profile in this example:

- 1 Create a new Connection profile:

```
admin> new connection dslpipe
```

- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Set the encapsulation type to PPP:

```
admin> set encapsulation-protocol=ppp
```

- 4 List the IP-Options submenu:

```
admin> list ip-options

[in CONNECTION/dslpipe:ip-options]
ip-routing-enabled = yes
vj-header-prediction = yes
remote-address = 0.0.0.0/0
local-address = 0.0.0.0/0
..
..
```

- 5 Set the IP address of the DSLPipe connecting to the DSLTNT:

```
admin> set remote-address=10.10.73.1/24
```

Configuring DSL Connections (DSLNT)

Sample DSL configurations

- 6 Verify that IP routing is enabled (the default) for this Connection profile:

```
admin> set ip-routing-enabled = yes
```
- 7 Verify that VJ header prediction is not enabled for this Connection profile:

```
admin> set vj-header-prediction = no
```
- 8 List the PPP-Options submenu:

```
admin> list .. ppp-options
```

```
[in CONNECTION/dslpipe:ppp-options]
send-auth-mode = no-ppp-auth
send-password = ""
recv-password = ""
link-compression = stac
mru = 1524
lqm = no
lqm-minimum-period = 600
lqm-maximum-period = 600
split-code-dot-user-enabled = no
```
- 9 Specify the authentication mode that the DSLTNT requests for the outgoing call:

```
admin> set send-auth-mode = pap-ppp-auth
```
- 10 Specify the password that the DSLTNT sends to the DSLPipe:

```
admin> set send-password = pap
```
- 11 Specify the password that the DSLTNT expects to receive from the DSLPipe:

```
admin> set recv-password = pap
```
- 12 List the Telco-Options submenu:

```
admin> list .. telco-options
```

```
[in CONNECTION/dslpipe:telco-options]
answer-originate = ans-and-orig
callback = no
call-type = off
nailed-groups = 1
ftl-caller = no
force-56kbps = no
data-service = 56k-clear
..
..
```
- 13 Specify the call type:

```
admin> set call-type= ftl
```
- 14 Specify the nailed group to use for this Connection profile:

```
admin> set nailed-groups = 73
```
- 15 Write the profile:

```
admin> write
```

Configuring the DSLPipe

To configure the DSLPipe in this example:

- 1 From the Main Edit menu, select Configure.

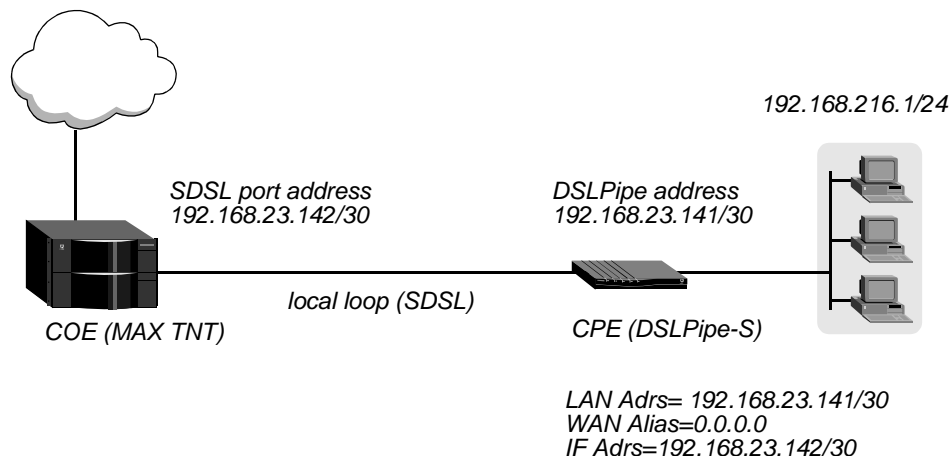
- 2 Specify the following values:
 - Chan Usage=**Leased/Unused**
 - My Name=**dslpipe**
 - My Addr=**10.10.73.1/24**
 - Rem Name=**max-tnt**
 - Rem Addr=**104.178.115.163/24**
 - Route=**IP**
 - Bridge=**No**
- 3 From the Main Edit menu, select Ethernet > Connections > max-tnt.
- 4 Specify the following values:
 - Active=**Yes**
 - Encaps=**PPP**
 - Bridge=**No**
 - Route IP=**Yes**
- 5 Open the Encaps Options submenu.
- 6 Specify the following values:
 - Send Auth=**PAP**
 - Send PW=**PAP**
 - Recv PW=**PAP**
 - Link Comp=**None**
 - VJ Comp=**No**
- 7 Open the Telco Options submenu.
- 8 Specify the following values:
 - Call Type=**Nailed**
 - Group=**1**
- 9 Exit the Connection profile and save your changes.

Sample SDSL Frame Relay configuration using numbered interfaces

This section describes a common SDSL application. In this example, the SDSL line is a leased connection over a single pair of wires, using Frame Relay as the transport protocol (see Figure 17-5). The example uses interface-based routing on a point-to-point link. Each side of the connection is assigned a unique address that applies only to the connection.

This example uses SDSL, but you can configure an ADSL connection similarly.

Figure 17-5. Example SDSL setup with interface-based routing



Configuring an SDSL connection requires the following general steps:

- Configuring the Connection profile
- Configuring an IP-Route profile
- Configuring the SDSL profile
- Configuring the Frame-Relay profile
- Configuring the DSLPipe-S

Configuring the Connection profile

To configure the Connection profile:

- 1 Create a new Connection profile:
`admin> new connection sdsl-pipeline`
- 2 Enable the profile:
`admin> set active=yes`
- 3 Specify the encapsulation type as Frame Relay:
`admin> set encapsulation-protocol=frame-relay`
- 4 List the IP-Options submenu:
`admin> list ip-options`
[in CONNECTION/sdsl-pipeline:ip-options]
ip-routing-enabled = yes
vj-header-prediction = yes
remote-address = 0.0.0.0/0
local-address = 0.0.0.0/0
..
..
- 5 Set the IP address of the DSLPipe-S connecting to the DSLTNT:
`admin> set remote-address=192.168.23.141/30`
- 6 Set the IP address of the DSLTNT SDSL port:

- ```
admin> set local-address=192.168.23.142/30
```
- 7 List the submenu for Frame Relay options:
- ```
admin> list .. fr-options

[in CONNECTION/sdsl-pipeline:fr-options]
frame-relay-profile = ""
dlci = 16
circuit-name = ""
fr-direct-enabled = no
fr-direct-profile = ""
fr-direct-dlci = 16
```
- 8 Link this Connection profile to the Frame-Relay profile you will create in the next section:
- ```
admin> set frame-relay-profile=fr-prof-1
```
- 9 Set the DLCI to the same value as the DSLPipe-S:
- ```
admin> set dlci=16
```
- 10 Open the Telco-Options subprofile:
- ```
admin> list .. telco-options

[in CONNECTION/sdsl-pipeline:telco-options]
answer-originate = ans-and-orig
callback = no
call-type = off
nailed-groups = 1
ft1-caller = no
force-56kbps = no
data-service = 56k-clear
..
..
```
- 11 Specify that the that the connection only uses nailed channels by setting Call-Type to FT1 (fractional T1):
- ```
admin> set call-type=ft1
```
- 12 Write the profile:
- ```
admin> write
```

### *Configuring the IP-Route profile*

Next, to properly route traffic to machines on the DSLPipe unit's LAN:

- 1 Create a new IP Routing profile:
- ```
admin> new ip-route sdsl-pipeline
```
- 2 Set the address to route equal to the Pipeline's LAN address:
- ```
admin> set dest-address=192.168.216.1/24
```
- 3 Set the gateway to the interface address assigned to the DSLPipe:
- ```
admin> set gateway-address=192.168.23.141
```
- 4 Write the profile:
- ```
admin> write
```

#### Configuring the SDSL profile

To configure the SDSL profile:

- 1 Read in the SDSL profile. For example, if the SDSL card is installed in slot 11 of shelf 1 and the remote DSLPipe-S is connected to port 1:

```
admin> read sdsl {1 11 1}
```

- 2 Enable the port:

```
admin> set enabled=yes
```

- 3 List the contents of the Line-Config profile:

```
admin> list line-config

[in SDSL/{ shelf-1 slot-11 1 }:line-config]
trunk-group = 0
nailed-group = 1
activation = static
call-route-info = { any-shelf any-slot 0 }
max-rate = 144000
unit-type = coe
```

- 4 Assign this port to a nailed group:

```
admin> set nailed-group=1
```

This nailed group points to the Frame-Relay profile you will create later. The nailed group must be unique for each active WAN interface.

- 5 Write the profile:

```
admin> write
```

#### Configuring the Frame-Relay profile

See the *APX 8000/MAX TNT/DSLNT Frame Relay Configuration Guide* for more information and examples.

To configure the Frame-Relay profile:

- 1 Create a new Frame-Relay profile:

```
admin> new frame-relay fr-prof-1
```

- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Assign the Frame-Relay profile to a nailed-up group:

```
admin> set nailed-up-group=1
```

This value must be the same as the SDSL nailed group number you configured in the SDSL profile. The nailed group must be unique for each active WAN interface.

- 4 Write the profile:

```
admin> write
```



## Configuring the DSLPipe-S

This section provides an example of configuring the SDSL Pipeline (DSLPipe-S). For complete information about configuring the DSLPipe-S, see the documentation that came with your Pipeline unit.

Before you configure the Pipeline, make sure of the following:

- The PC connected to the Pipeline has an IP address on the same subnet as the Pipeline.
- The IP address of the Pipeline is configured as the default gateway for the PC.

To configure the Pipeline:

- 1 From the Main Edit menu, select Configure.
- 2 Specify the following values:
  - Chan Usage=**Leased/Unused**
  - My Name=**sdsl-pipeline**
  - My Addr=**192.168.216.1/24**
  - Rem Name=**max-tnt**
  - Rem Addr=**192.168.23.142/30**
  - Route=**IP**
- 3 Exit and save the Configure profile.
- 4 From the Main Edit menu, select Ethernet > Connections > max-tnt.
- 5 Specify the following values:
  - Active=**Yes**
  - Encaps=**FR**
  - Route IP=**Yes**
- 6 Open the Encaps Options submenu.
- 7 Specify the following values:
  - FR Prof=**Frame Relay**
  - DLCI=**16**
- 8 Open the IP options submenu.
- 9 Specify the following values:
  - LAN Adrs=**192.168.23.142/30**
  - WAN Alias=**0.0.0.0**
  - IF Adrs=**192.168.23.141/30**
- 10 Exit the Connection profile and save your changes.

Next, set up the Frame-Relay profile.

- 1 Open the Ethernet > Frame Relay > Frame Relay profile.

- 2 Specify the following values:
  - Name=**Frame Relay**
  - Active=**Yes**
  - Call Type=**Nailed**
- 3 If your Pipeline supports it, set LinkUp to Yes:
  - LinkUp=**Yes**

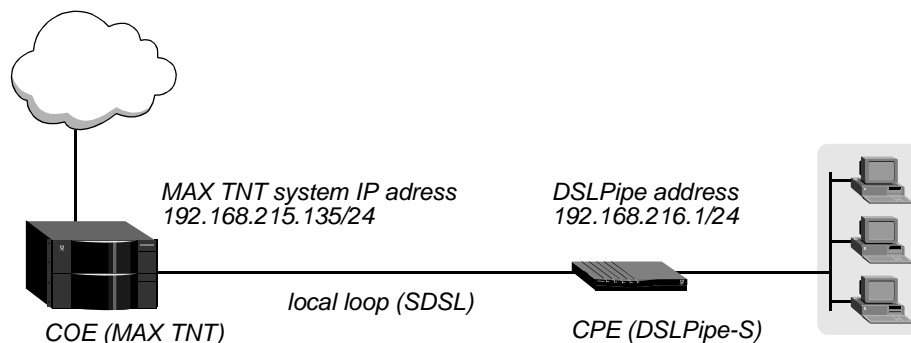
Note that this parameter does not appear in recent versions of Pipeline software.
- 4 Exit the Frame-Relay profile and save your changes.

## Sample SDSL Frame Relay configuration using system-based routing

This section describes a common SDSL application. In this example, the SDSL line is a leased connection over a single pair of wires, using Frame Relay as the transport protocol (see Figure 17-6). The example uses system-based routing. In system-based routing each system has an IP address. The system routes traffic based on the destination address in packets and the next-hop system.

This example uses SDSL, but you can configure an ADSL connection similarly.

*Figure 17-6. Example SDSL setup with system-based routing*



Configuring an SDSL connection requires the following general steps:

- Configuring the Connection profile
- Configuring the SDSL profile
- Configuring the Frame-Relay profile
- Configuring the DSLPipe-S

### Configuring the Connection profile

To configure the Connection profile:

- 1 Create a new Connection profile:

```
admin> new connection sdsl-pipeline
```
- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Specify the encapsulation type as Frame Relay:

```
admin> set encapsulation-protocol=frame-relay
```

- 4 List the IP-Options submenu:

```
admin> list ip-options

[in CONNECTION/sdsl-pipeline:ip-options]
ip-routing-enabled = yes
vj-header-prediction = yes
remote-address = 0.0.0.0/0
local-address = 0.0.0.0/0
....
```

- 5 Set the IP address of the DSLPipe-S connecting to the DSLTNT:

```
admin> set remote-address=192.168.216.1/24
```

- 6 List the submenu for Frame Relay options:

```
admin> list .. fr-options

[in CONNECTION/sdsl-pipeline:fr-options]
frame-relay-profile = ""
dlci = 16
circuit-name = ""
fr-direct-enabled = no
fr-direct-profile = ""
fr-direct-dlci = 16
```

- 7 Link this Connection profile to the Frame-Relay profile you will create in the next section:

```
admin> set frame-relay-profile=fr-prof-1
```

- 8 Set the DLCI to the same value as the DSLPipe-S:

```
admin> set dlci=16
```

- 9 Open the Telco-Options subprofile:

```
admin> list .. telco-options

[in CONNECTION/sdsl-pipeline:telco-options]
answer-originate = ans-and-orig
callback = no
call-type = off
nailed-groups = 1
ft1-caller = no
force-56kbps = no
data-service = 56k-clear
..
..
```

- 10 Specify that the connection only uses nailed channels by setting Call-Type to FT1 (fractional T1):

```
admin> set call-type=ft1
```

- 11 Write the profile:

```
admin> write
```

## *Configuring the SDSL profile*

To configure the SDSL profile:

- 1 Read in the SDSL profile. For example, if the SDSL card is installed in slot 11 of shelf 1 and the remote DSLPipe-S is connected to port 1:

```
admin> read sdsl {1 11 1}
```

- 2 Enable the port:

```
admin> set enabled=yes
```

- 3 List the contents of the Line-Config profile:

```
admin> list line-config

[in SDSL/{ shelf-1 slot-11 1 }:line-config]
trunk-group = 0
nailed-group = 1
activation = static
call-route-info = { any-shelf any-slot 0 }
max-rate = 144000
unit-type = coe
```

- 4 Assign this port to a nailed group:

```
admin> set nailed-group=1
```

This nailed group points to the Frame-Relay profile you will create later. The nailed group must be unique for each active WAN interface.

- 5 Write the profile:

```
admin> write
```

## *Configuring the Frame-Relay profile*

To configure the Frame-Relay profile:

- 1 Create a new Frame-Relay profile:

```
admin> new frame-relay fr-prof-1
```

- 2 Enable the profile:

```
admin> set active=yes
```

- 3 Assign the Frame-Relay profile to a nailed-up group:

```
admin> set nailed-up-group=1
```

This must be the same as the SDSL nailed group number you configured in the SDSL profile. The nailed group must be unique for each active WAN interface.

- 4 Write the profile:

```
admin> write
```

## *Configuring the DSLPipe-S*

This section provides an example of configuring the SDSL Pipeline (DSLPipe-S). For complete information about configuring the DSLPipe-S, see the documentation that came with your Pipeline unit.

Before you configure the Pipeline, make sure of the following:

- The PC connected to the Pipeline has an IP address on the same subnet as the Pipeline.
- The IP address of the Pipeline is configured as the default gateway for the PC.

To configure the Pipeline:

- 1 From the Main Edit menu, select Configure.
- 2 Specify the following values:
  - Chan Usage=**Leased/Unused**
  - My Name=**sdsl-pipeline**
  - My Addr=**192.168.216.1/24**
  - Rem Name=**max-tnt**
  - Rem Addr=**192.168.215.135/24**
  - Route=**IP**
- 3 Exit and save the Configure profile.
- 4 From the Main Edit menu, select Ethernet > Connections > max-tnt.
- 5 Specify the following values:
  - Active=**Yes**
  - Encaps=**FR**
  - Route IP=**Yes**
- 6 Open the Encaps Options submenu.
- 7 Specify the following values:
  - FR Prof=**Frame Relay**
  - DLCI=**16**
- 8 Exit the Connection profile and save your changes.

Next, set up the Frame-Relay profile.

- 1 Open the Ethernet > Frame Relay > Frame Relay profile.
- 2 Specify the following values:
  - Name=**Frame Relay**
  - Active=**Yes**
  - Call Type=**Nailed**
- 3 If your Pipeline supports it, set LinkUp to Yes:
  - LinkUp=**Yes**

Note that this parameter does not appear in recent versions of Pipeline software.

- 4 Exit the Frame-Relay profile and save your changes.



# Signaling System 7 (SS7)

# 18

|                                                        |       |
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## Introduction to SS7

Signaling System 7 (SS7) is an internationally standardized general-purpose common-channel signaling system designed for use over a variety of digital circuit-switched networks. At the physical layer, it uses T1, T3, or E1 for data traffic and separate time-division multiplexing (TDM) circuits for signaling information.

The following two methods of integration with an SS7 network are supported, each of which requires a separate software license:

- Access SS7 Gateway Control Protocol (ASGCP). This method of integration enables the TAOS unit to terminate data calls in an SS7 network. The signaling gateway must be ICD for softswitch (formerly ASG). ICD stands for Internet Call Diversion.
- IP Device Control (IPDC). IPDC is a third-party proprietary protocol. This method of integration enables the TAOS unit to terminate both voice and data calls. The signaling gateway can be ICD for softswitch or Lucent Softswitch.

Table 18-1 shows the protocols supported by these signaling gateway platforms.

*Table 18-1. Signaling gateway platforms and protocol support*

| Platform                          | IPDC 0.12 | ASGCP (Q.931+) |
|-----------------------------------|-----------|----------------|
| ICD for softswitch (formerly ASG) | Supported | Supported      |
| Lucent Softswitch                 | Supported | Not supported  |

## System requirements for SS7 operations

A TAOS unit configured for SS7 in communication with an SS7 signaling gateway is a service switching point (SSP). To operate in this capacity, the TAOS unit must have the following equipment and licenses:

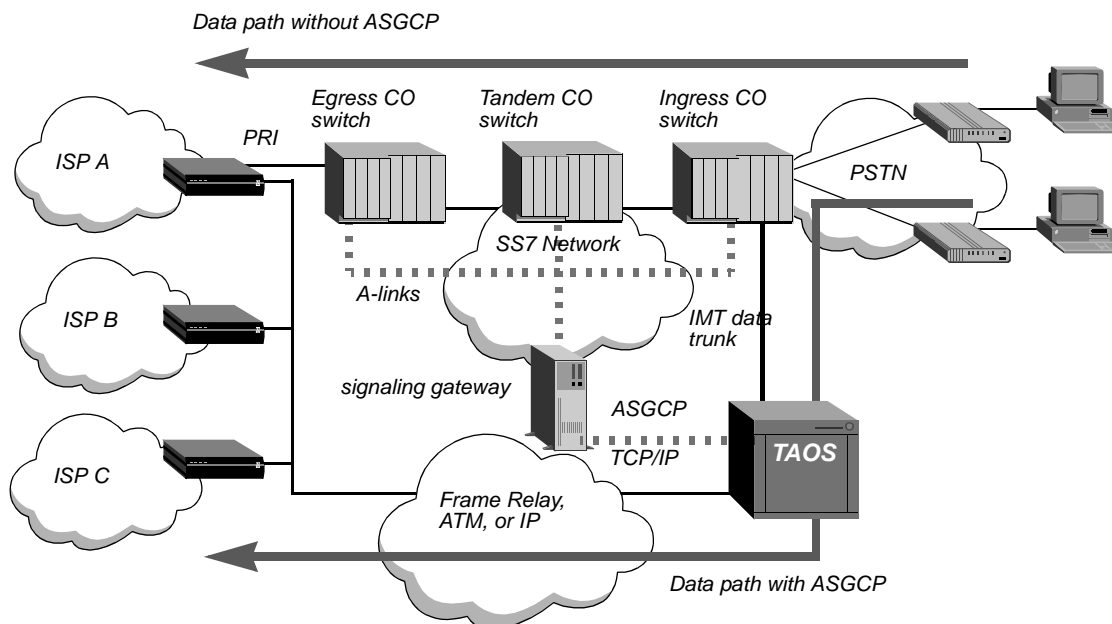
- SS7 software license, either for ASGCP or IPDC
- Sufficient T1, T3, or E1 trunks
- Sufficient modem or Hybrid Access (HDLC) cards (or both) to terminate data calls
- One or more Ethernet cards (recommended to offload the shelf controller)

If the system is a MAX TNT unit and will operate as a MultiVoice gateway in an SS7 environment, a MultiVoice software license must also be enabled and one or more MultiDSP cards must be installed to enable the system to terminate voice calls. For details about MultiVoice, see the *MultiVoice for MAX TNT Configuration Guide*.

### TAOS unit as terminator of data calls in an SS7 network

With the ASGCP license, TAOS units can decrease congestion on the Public Switched Telephone Network (PSTN) caused by users connecting to the Internet. An example of a TAOS unit being used for this purpose is shown in Figure 18-1.

*Figure 18-1. TAOS terminating data calls in an SS7 network*



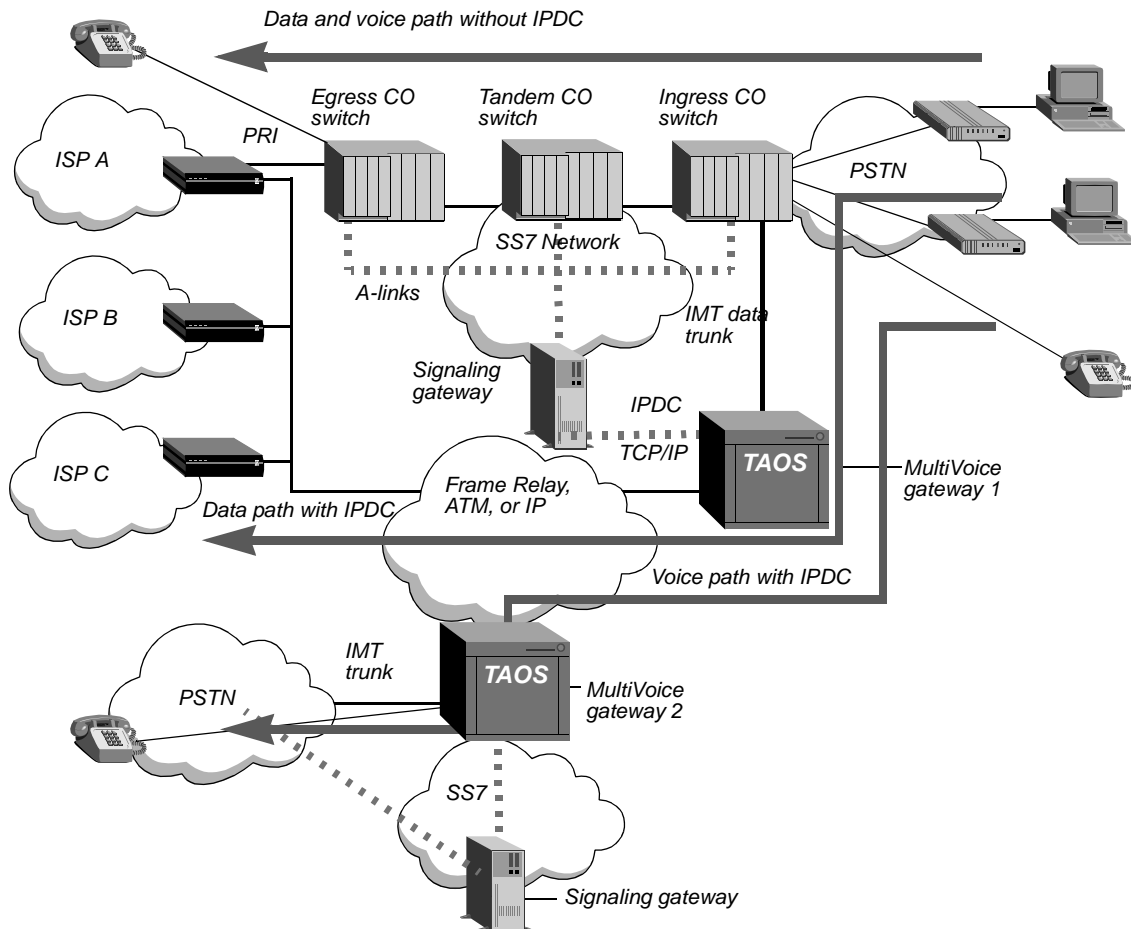
The TAOS unit is connected to the entry (ingress) central office (CO) switch via intermachine trunks (IMTs) and to a signaling gateway by means of dual-link (primary and secondary) TCP/IP links. Each CO switch is a service switching point (SSP). The combination of a TAOS unit and signaling gateway is also an SSP. The signaling gateway is connected to the SS7 network by access links (A-links). The signaling gateway and the TAOS unit together act as a switch that routes calls intended for ISPs directly to the TAOS unit, thus avoiding the PSTN tandem or transit switches and interoffice trunks.



## TAOS unit as terminator of voice and data calls in an SS7 network

With the IPDC license, the TAOS unit can communicate with the signaling gateway by means of IPDC. IPDC enables the TAOS unit to terminate voice or data calls. An example of TAOS units being used both for Internet call diversion (data) and Voice over IP (VoIP) is shown in Figure 18-2.

Figure 18-2. TAOS unit terminating voice and data calls in an SS7 network



Connection to the SS7 network is achieved through a signaling gateway. This gateway provides a bridge to the SS7 network and performs service switching point functions such as initiating and managing call setup and release, and executing call routing. IPDC must be supported by both the signaling gateway and the TAOS unit.

The signaling gateway uses the IPDC protocol to convert the SS7 signaling information and call data from the PSTN into IPDC packets, which are sent to the TAOS unit. In addition, the gateway uses IPDC to convert IPDC packets received from a TAOS unit into SS7 format before sending the call to the PSTN.

Before sending call data across the IP network, the TAOS unit uses IPDC to extract TDM and IP routing instructions from the IPDC packets received from the signaling gateway. The far-end TAOS unit then forwards IPDC packets to a signaling gateway, which converts them back into SS7 messages before the call is connected.

See *MultiVoice for MAX TNT Configuration Guide* for a more detailed description of how VoIP calls are processed by IPDC.

## **Interface between a signaling gateway and TAOS unit**

TCP/IP is the transport service used to carry control messages between a signaling gateway and the TAOS unit. The data delivery layer (DDL) uses a TCP/IP socket on both the signaling gateway and TAOS unit. On the signaling gateway side, the DDL is the server that listens for the socket connection and keeps track of the mapping between a TAOS unit and its socket. On the TAOS unit side, the DDL is the client that initiates a socket connection and handles connection establishment, connection recovery, and link selection.

## **Incoming calls**

The ingress central office (CO) switch (see Figure 18-1) processes the incoming call based on the called number, then identifies the TAOS unit as the destination for the call. The SS7 network sends an initial address message (IAM) to the signaling gateway. The signaling gateway informs the TAOS unit that a call will be coming in on one of the IMT channels from the CO switch. The message from the CO switch contains the calling and called party number, the circuit identification code (CIC), and the destination point code (DPC). The signaling gateway sends an address complete message (ACM) to the SS7 network acknowledging that it has received the relevant information to route the call.

The signaling gateway then sends a call origination message to the TAOS unit to establish a path between the ingress switch and the TAOS unit. The TAOS unit sets up the path and then sends an answer message to the signaling gateway so that the signaling gateway can make the proper updates to its resource management database. For a T1 or T3 network, the signaling gateway then sends an answer message to the SS7 network.

Once the path is set up, the TAOS unit accepts the call, off-loading the Internet call from the PSTN to the data network. The data network used to off-load the call can be a Frame Relay, ATM, or IP network.

## **Continuity tests**

A continuity test can be performed at the time of call setup or during testing to verify that the physical link between the CO switch and the TAOS unit is available. The CO switch informs the signaling gateway, which then informs the TAOS unit that it will conduct a continuity test on the circuit. During a call continuity test, the CO switch sends a tone through the physical path to the TAOS unit and receives a tone back from the TAOS unit indicating the continuity of the path.

## **Configuring an SS7 signaling gateway**

The signaling gateway and TAOS unit communicate over a TCP/IP link. The signaling interface can be a single or dual TCP connection between the TAOS unit and signaling gateway. When the interface initializes, it opens TCP connections to the specified addresses and ports of the signaling gateway. The TAOS unit keeps the TCP connections open as long as the unit is operating and the signaling interface is enabled.

Settings in the SS7-Gateway profile configure the signaling interface. The TAOS unit resets the signaling link whenever changes are written to the profile.

Following are the parameters (shown with default settings) for configuring the signaling interface:

```
[in SS7-GATEWAY]
enabled = no
control-protocol = asgcp
primary-ip-address = 0.0.0.0
primary-tcp-port = 0
secondary-ip-address = 0.0.0.0
secondary-tcp-port = 0
bay-id = ""
system-type = IASCTNT1B
transport-options = { 0 1000 3000 30000 7 6 no }
use-system-ip-address-as-source = yes
```

| Parameter                                  | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
|--------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Enabled                                    | Enable/disable the interface. When this parameter is set to <code>no</code> (the default), the interface is disabled. When it is set to <code>yes</code> , the interface is enabled if the Primary-IP-Address and Primary-TCP-Port also have valid values. Changing the setting from <code>yes</code> to <code>no</code> closes the signaling links but does not disconnect active SS7 calls.                                                                                                                                                                       |
| Control-Protocol                           | Control protocol. The <code>asgcp</code> setting enables the unit to terminate data calls by using ASGCP. The <code>ipdc-0.x</code> (XCOM/Level 3 IPDC) setting enables the unit to terminate voice and data using IPDC. If only one SS7 license is enabled, the parameter defaults to that control protocol ( <code>asgcp</code> or <code>ipdc-0.x</code> ) and cannot be modified. If both licenses are enabled, the parameter defaults to <code>asgcp</code> . See “Specifying the SS7 control protocol” on page 18-6 for more information about this parameter. |
| Primary-IP-Address<br>Primary-TCP-Port     | IP address and TCP port to use for communication with the primary signaling gateway. These settings are required for SS7 operations.                                                                                                                                                                                                                                                                                                                                                                                                                                |
| Secondary-IP-Address<br>Secondary-TCP-Port | IP address and TCP port to use for communication with a secondary signaling gateway. These settings are optional. If specified, the secondary signaling gateway is used only when the primary gateway is unavailable. The primary and secondary address and port configurations can point to two Ethernet interfaces of the same signaling gateway.                                                                                                                                                                                                                 |
| Bay-ID                                     | This parameter does not apply when Control-Protocol is set to <code>asgcp</code> . When Control-Protocol is set to <code>ipdc-0.x</code> , the system sends its value as an ASCII string to the media gateway controller in the device registration message. The TAOS unit does not interpret the value. Interpretation on the signaling gateway is gateway dependent.                                                                                                                                                                                              |

| Parameter                       | Specifies                                                                                                                                                                                                                                                                                                                                                              |
|---------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| System-Type                     | This parameter does not apply when Control-Protocol is set to <code>asgcp</code> . When Control-Protocol is set to <code>ipdc-0.x</code> , the system sends its value as an ASCII string to the media gateway controller in the device registration message. The TAOS unit does not interpret the value. Interpretation on the signaling gateway is gateway dependent. |
| Transport-Options               | The Transport-Options subprofile contains settings for changing the operation of SS7 DDL timers. See “Configuring transport-layer options” on page 18-6.                                                                                                                                                                                                               |
| Use-System-IP-Address-As-Source | Enable/disable use of the system address as the source address for packets generated by the TAOS unit. See “System IP address considerations” on page 18-7.                                                                                                                                                                                                            |

## Specifying the SS7 control protocol

With the appropriate software license, TAOS supports either ASGCP and IPDC 0.12 control protocol. If only one of the possible control protocols (`asgcp` or `ipdc-0.x`) is licensed on the TAOS unit, the Control-Protocol parameter defaults to the licensed protocol and cannot be modified. However, if both protocols are licensed, the parameter defaults to `asgcp`. Because of this default and because the TAOS unit does not store unmodified profile items in NVRAM, the setting can be modified unintentionally when you upgrade to new software or enable a new license to support a second control protocol. For this reason, Lucent recommends that you verify the setting after upgrading. If the proper protocol is not specified, change the setting and then reset the unit.

Although the control protocol is configurable in real time, you must reset the system to begin using the new protocol. After the TAOS unit is reset, it establishes a new TCP link to the signaling gateway and begins communicating with it using the specified control protocol.

## Configuring transport-layer options

Administrators occasionally need to change the duration of various SS7 DDL timers to fine-tune a signaling link. For example, you might want to change timeouts when integrating a TAOS unit with existing signaling gateways. The following parameters, shown with default values, are used to set TAOS time intervals for waiting and responding to the various signaling link processes:

```
[in SS7-GATEWAY:transport-options]
device-id = 0
t1-duration = 1000
t2-duration = 3000
t3-duration = 30000
window-size = 7
ack-threshold = 6
heart-beat = no
```

| Parameter     | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                      |
|---------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Device-ID     | Logical SS7 command control device where these values apply. Currently, the settings in this profile apply only to the TAOS unit's operations. <i>This parameter is currently not used.</i>                                                                                                                                                                                                                                                    |
| T1-Duration   | Value of the acknowledgement (ACK) delay timer in milliseconds. This timer specifies the maximum delay for an acknowledgement when an information frame (I-frame) is received. The default value is 1000 (1 second). The value must be less than the T2 duration timer specified on the signaling gateway. Valid values range from 0 to 2147483647.                                                                                            |
| T2-Duration   | Value of the transmission time-out timer in milliseconds. This timer specifies how long this endpoint must wait for an acknowledgement to a heartbeat frame. The default value is 3000 (3 seconds). The value must be greater than the T1 duration timer on the signaling gateway. Valid values range from 0 to 2147483647.                                                                                                                    |
| T3-Duration   | Value of the persistent error timer in milliseconds. This timer specifies the maximum duration of attempts to reestablish a link before the transport layer flushes the data queues and sends an error indication up. Default value is 30000 (30 seconds). Valid values range from 0 to 2147483647.                                                                                                                                            |
| Window-Size   | Maximum number of sequentially numbered data packets that can be sent while pending acknowledgement at any given time. Default value is 7. Valid values range from 1 to 63.                                                                                                                                                                                                                                                                    |
| Ack-Threshold | Threshold for triggering an acknowledgement (ACK) while receiving data packets. As soon as the specified number of packets is received, the TAOS unit sends an ACK back regardless of the value of its T1 timer. The value of this parameter must not be greater than the window size. Default value is 6. Valid values range from 1 to 63.                                                                                                    |
| Heartbeat     | Enable/disable detection of a physical link failure, such as disconnection of a cable or failure of the signaling gateway. When the parameter is set to <i>yes</i> , the TAOS unit periodically sends out heartbeat frames to the signaling gateway and waits for an acknowledgement. If it does not receive an acknowledgement within the number of milliseconds specified in its T2-Duration timer, the TAOS unit resets the signaling link. |

## System IP address considerations

The System-IP-Addr parameter of the IP-Global profile specifies the source address of all packets generated by the system, such as the connection request packets sent to a signaling gateway to establish communication. When the Use-System-IP-Address-As-Source parameter is set to *yes* (the default), the TAOS unit uses the system address as its source address in the packets it sends to the signaling gateway.

For some sites, administrative policy or other constraints introduce a requirement to use the system address for some purposes, but to use a separate source address for communication with the signaling gateway. For example, although a site might require a certain system address for compatibility with other routers, this requirement might cause an address space conflict, or might cause delays and time-outs in the receipt of acknowledgements from signaling gateways. Or, a site might decide to separate the signaling control network from the Internet for security purposes.

To enable sites to integrate TAOS units into their infrastructure and at the same time communicate efficiently with signaling gateways, the following parameter (shown with its default value) was introduced:

```
[in SS7-GATEWAY]
use-system-ip-address-as-source = yes
```

When this parameter is set to no, the TAOS unit does not use the system address as its source address for signaling packets. Instead, it uses the IP address of the Ethernet interface on which the signaling packets are sent. When the parameter is set to yes, the TAOS unit uses the same system address for signaling packets as for all other packets generated by the system.

## Example of a basic configuration

The following commands configure an SS7-Gateway profile for a single TCP connection to a signaling gateway running IPDC:

```
admin> read ss7-gateway
SS7-GATEWAY read

admin> set enabled = yes

admin> set primary-ip-address = 1.1.1.1

admin> set primary-tcp-port = 5000

admin> write
SS7-GATEWAY written
```

**Note:** For the link to become active, the signaling gateway must have a matching entry for the TAOS unit. For information about configuring the signaling gateway, see the documentation that came with the unit.

## T1 lines as SS7 data trunks

T1 lines:signaling system 7 (SS7) data trunks;To configure T1 lines for SS7, you must set the following parameters, shown with sample settings:

```
[in T1/{ shelf-1 slot-1 7 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing
```

```
[in T1/{ shelf-1 slot-1 7 }:line-interface:channel-config:24]
channel-usage = switched-channel
```

| Parameter              | Usage for SS7 data trunks                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                         |
|------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Signaling-Mode         | <p>For an SS7 data trunk, which carries no signaling, this parameter can be set to either of the following values. The setting registers the line with the signaling gateway and allows the gateway to take control of the line and its calls.</p> <p><code>ss7-data-trunk</code> causes the unit to provide clear 64Kbps SS7 data trunk support. If any of the PSTN switches you are using is a 1AESS switch, which uses robbed-bit signaling, this setting can sometimes cause that switch to receive fluctuating A/B bit status. This condition might ultimately force the line out of service, unless you disable robbed-bit signaling on the 1AESS switch.</p> <p><code>ss7-robbed-bit</code> causes the TAOS unit to send a steady A/B bit status on the SS7 data trunk, which eliminates the need to disable robbed-bit signaling on the 1AESS switch.</p> |
| Incoming-Call-Handling | <p>Specifies how the TAOS unit processes incoming calls on this line. For SS7 data trunks, the parameter must be set to <code>internal-processing</code> in this release. The <code>ss7-gateway-processing</code> setting for passing incoming call requests to an external signaling gateway is currently not supported.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
| Channel-Usage          | <p>T1 lines typically use channel 24 for signaling. For SS7 data trunks, the Channel-Usage setting for channel 24 must be <code>switched-channel</code>.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                      |

### *Example of configuring a T3 card for SS7 data*

To configure lines of a T3 card as SS7 data trunks, you must first configure the T3 profile as in the following example:

```
admin> read t3 {1 1 1}
T3/{ shelf-1 slot-1 1 } read
admin> set enabled = yes
admin> set frame-type = ml3
admin> set line-length = 0-225
admin> write
T3/{ shelf-1 slot-1 1 } written
```

After configuring the T3 line, configure the individual T1 lines that constitute the T3 line as explained in the next section.

### *Example of configuring a T1 data trunk*

The following commands configure a T1 line as an SS7 data trunk, enabling the signaling gateway to control the line:

```
admin> read t1 {1 1 7}
T1/{ shelf-1 slot-1 7 } read
```

```
admin> set line-interface enabled = yes

admin> set line-interface signaling-mode = ss7-data-trunk

admin> set line-interface incoming-call-handling = internal-processing

admin> set line-interface channel-config 24 channel-usage = switched

admin> write
T1/{ shelf-1 slot-1 7 } written
```

## E1 lines as SS7 data trunks

Configuring the E1 SS7 data trunks is very similar to configuring T1 data trunks. To configure E1 lines for SS7, you must set the following parameters in an E1 profile, shown with sample settings:

```
[in E1/{ shelf-1 slot-10 1 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing

[in E1/{ shelf-1 slot-10 1 }:line-interface:channel-config[17]]
channel-usage = switched-channel
```

| Parameter              | Usage for SS7 data trunks                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Signaling-Mode         | <p>For an SS7 data trunk, which carries no signaling, this parameter can be set to either of the following values. The setting registers the line with the signaling gateway and allows the gateway to take control of the line and its calls.</p> <p><code>ss7-data-trunk</code> causes the TAOS unit to provide clear 64Kbps SS7 data trunk support. If any of the PSTN switches you are using is a 1AESS switch, which uses robbed-bit signaling, this setting can sometimes cause that switch to receive fluctuating A/B bit status. This condition might ultimately force the line out of service, unless you disable robbed-bit signaling on the 1AESS switch.</p> <p><code>ss7-robbed-bit</code> causes the TAOS unit to send a steady A/B bit status on the SS7 data trunk, which eliminates the need to disable robbed-bit signaling on the 1AESS switch.</p> |
| Incoming-Call-Handling | <p>Specifies how the TAOS unit processes incoming calls on this line. For SS7 data trunks, the parameter must be set to <code>internal-processing</code> in this release. The <code>ss7-gateway-processing</code> setting for passing incoming call requests to an external signaling gateway is currently not supported.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |
| Channel-Usage          | <p>In the TAOS unit, the channel-config index begins with 1 (not 0), so E1 lines typically use channel 17 for signaling. For SS7 data trunks, change the default Channel-Usage setting for channel 17 from <code>d-channel</code> to <code>switched-channel</code>.</p>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                |



For example, the following commands configure an E1 line as an SS7 data trunk, enabling the signaling gateway to control the line:

```
admin> read e1 {1 10 1}
E1/{ shelf-1 slot-10 1 } read

admin> set line-interface enabled = yes

admin> set line-interface signaling-mode = ss7-data-trunk

admin> set line-interface incoming-call-handling = internal-processing

admin> set line-interface channel-config 17 channel-usage = switched

admin> write
E1/{ shelf-1 slot-10 1 } written
```

## V.110 bearer capability for SS7 calls using IPDC

TAOS supports V.110 bearer capability for SS7 calls using IPDC. This feature enables SS7 call routing across V.110 interfaces in the TAOS unit. Use of this capability is controlled via IPDC messages from the signaling gateway.

## SS7 link establishment timer

TAOS supports a T5 timer that automatically enables link connection and reconnection requests to the signaling gateway to occur at random intervals. The timer can help prevent the signaling gateway from receiving many link connect requests within a short period of time, especially when the signaling gateway is connected with many TAOS units.

After its link to the gateway is disconnected, the TAOS unit initializes the T5 timer with a random value between 0 and 6 seconds and attempts a connection when the timer expires. After each failed connection attempt, the TAOS unit increases the T5 time-out value by 1 second until it reaches 20 seconds. The timer remains at 20 seconds for subsequent connection attempts.

The TAOS unit resets the T5 timer as soon as the link is active.

## Two-wire continuity check on T1 and E1 lines

TAOS units support a 4-wire-only continuity check as defined in Q.724 Sections 7 and 8, ANSI T1.113.4 Annex B, GR-246-CORE Annex B on both T1 and E1 lines. The 4-wire continuity check requires one end of a line to place a channel into loopback state while the other end sends a tone. The check concludes successfully if the tone sent on the outgoing path is received on the return path within acceptable transmission and timing limits. The 4-wire check procedure cannot detect potential inadvertent loops in the line path or in line facilities, and cannot be used when the other exchange is analog. For these reasons, the procedure known as 2-wire continuity check is recommended by the International Telecommunications Union Telecommunication Standardization Sector (ITU-T), which carries out the operations of the former Consultative Committee for International Telephone and Telegraph (CCITT).

TAOS supports both incoming and outgoing 2-wire continuity checks for T1 lines only. You can select the type of check to perform on a

per-line basis. Both the native 2-wire continuity check (GR-246-CORE Section B.2) and 4-wire-to-2-wire emulation (GR-246-CORE Section B.3) are supported.

**Note:** Outgoing continuity tests are supported only on T1 and T3 cards. E1 cards support receipt of 4-wire continuity check requests only, and cannot originate continuity tests.

The SS7-Continuity subprofile in the T1 profile allows you to specify the type of incoming and outgoing continuity checks to perform for all channels on a line. Both ends of the connection must agree on the continuity check to be used for the line. Following are the relevant parameters, shown with default values:

```
[in T1/{ shelf-1 slot-1 1 }:line-interface:ss7-continuity]
incoming-procedure = loopback
outgoing-procedure = single-tone-2010
```

| Parameter          | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                    |
|--------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Incoming-Procedure | Loopback or transponder test mode. The <code>loopback</code> setting (the default) places the channel into loopback mode during the continuity test. This mode must be used if the line is provisioned for an incoming 4-wire continuity test. The <code>transponder</code> setting places the channel into Tone Transponder mode during the continuity test. In this mode, the channel can detect two tones: 2010Hz and 1780Hz. When either tone is detected, the other one is returned. This mode should be used for lines provisioned for incoming 2-wire and 4-wire-to-2-wire continuity checks.                                                                                                                                                                                                                                                                                                                                                         |
| Outgoing-Procedure | Type of continuity check. With the <code>single-tone-2010</code> setting (the default), the TAOS unit sends a 2010Hz tone and expects to receive a 2010Hz tone in return. This procedure is generally known as a 4-wire continuity check.<br><br>With the <code>send-2010-expect-1780</code> setting, the TAOS unit sends a 2010Hz tone and expects to receive 1780Hz tone in return. This procedure is generally known as a 2-wire continuity check.<br><br>With the <code>send-1780-expect-2010</code> setting, the TAOS unit sends a 1780Hz tone and expects to receive a 2010Hz tone in return. This procedure is generally known as a 4-wire to 2-wire continuity check.<br><br>If you change the type of a continuity check, the new type is used for new continuity check requests on the line as soon as the line profile is saved. Existing check-loops that are already active on the line are not modified or canceled when the profile is saved. |

The type of the continuity check procedure to be used is determined by line provisioning and is agreed upon by the connecting exchanges. SS7 signaling procedures used for continuity check (Q.764 Section G.3, ANSI T1.113.4 Section 2.1.6) are the same for both 4-wire and 2-wire circuits, but the behavior of trunk termination devices is different.

The native 2-wire continuity check procedure requires that the loopback be replaced by a transponder and that a 1780Hz  $\pm$  20Hz tone be used in the return direction.

TAOS also supports the 4-wire-to-2-wire continuity check, with the following requirements: The exchange that terminates 4 wires must use a transmitting frequency of  $1780 \pm 20\text{Hz}$  and a receiving frequency  $2010 \pm 30\text{Hz}$ . The exchange that terminates the 2 wires must use a transmitting frequency of  $2010 \pm 8\text{Hz}$  and a receiving frequency of  $1780 \pm 30\text{Hz}$ .

## Outgoing continuity tests on T1 and T3

TAOS units support incoming continuity tests on both T1 and E1 lines. During these tests, the telephone switch requests that the TAOS unit put a DS0 channel into a loopback and then generates a 2010Hz tone. If the switch receives the tone in return, the continuity test is successful.

TAOS also supports outgoing call continuity tests on T1 and T3 cards. For outgoing continuity, the switch puts a DS0 into a loopback and the TAOS unit generates a 2010Hz tone. If the TAOS unit receives the tone in return, the continuity test is successful. Note that all the setup and signaling required to coordinate a continuity test is handled by the signaling gateway via SS7.

**Note:** Outgoing continuity tests are supported only on T1 and T3 cards. E1 cards support receipt of 4-wire continuity check requests only, and cannot originate continuity tests.

## Digital milliwatt tone support on T1 and T3

T1 and T3 cards generate the 1000Hz digital milliwatt (DMW) tone. The SS7 switch sends a digital milliwatt tone request to the TAOS unit over IPDC and uses the tone that the TAOS unit generates in special test calls to measure the line distortion and attenuation in the telephone network.

## Analog milliwatt tone and variable tone support

IPDC Tone-Type and Tone-String tags enable the IPDC Specify Tone (STN) message to generate the analog milliwatt tones. When the TAOS unit receives a message from the signaling gateway specifying these tags, it responds with the appropriate tone type or tone string. When the signaling gateway specifies a variable tone, it details the tone in the IPDC Tone-String message tag, which uses the following format:

*"frequency1, frequency2, amplitude, duration"*

| Element    | Description                                                                                                                                                                                                                                                                                                     |
|------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Frequency1 | First frequency of the dual tone. This value can range from 1 to 3999 and has an accuracy of $\pm 1\text{Hz}$ .                                                                                                                                                                                                 |
| Frequency2 | The second frequency of the dual tone. This value can range from 0 for single tones to 3999 and has an accuracy of $\pm 1\text{Hz}$ .                                                                                                                                                                           |
| Amplitude  | Amplitude of the tone. If this value is in the range of 4 through 32767, it is an absolute value. A value in the range from -49 through 2 is a decibel level. The following relationship exists between decibel levels and absolute values:<br>$\text{dBm0} = 20 * \log_{10} (\text{absolute value} / 22748.4)$ |

| Element  | Description                                                                                                                                                                      |
|----------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Duration | Duration of the tone in milliseconds. This value can range from 0 to 2631. If the duration is 0, a tone will be played continuously until it is stopped by a second STN command. |

For example, the following string defines a 1004hz tone at 22748 amplitude for 1 second:

```
"1004, 0, 22748, 1000"
```

The following string defines a dual tone with frequency 697Hz and 1477Hz at 14567 amplitude for 2 seconds:

```
"697, 1477, 14567, 2000"
```

The following string defines a 2050, -3dBm0 tone played continuously until stopped by a second STN message:

```
"2050, 0, -3, 0"
```

## Reporting VoIP call statistics

A TAOS unit operating as a network access server (NAS) with a signaling gateway can report VoIP call statistics in the output of the NAS messaging interface. IPDC VoIP call statistics are reported once a call is cleared. The source that originates call clearing can be either the signaling gateway or the TAOS unit.

### *When the unit reports VoIP statistics*

IPDC 0.12 statistics tags are reported when the signaling gateway or the TAOS unit clears calls under the following conditions:

- When the access server initiates a call teardown using an RCR message.
- For packet-based calls when the access server acknowledges a call teardown using an ACR message

The TAOS unit reports the following VoIP statistics, as defined by IPDC 0.12:

- Number of Real-Time Protocol (RTP) audio packets sent and received by the TAOS unit.
- Number of RTP audio packets that failed to reach the TAOS unit as determined by missed sequence numbers.
- Number of audio bytes in the RTP payload sent by the TAOS unit.
- Number of audio bytes received in the RTP payload that failed to reach the TAOS unit. Because the number of bytes per packet is variable, this value can only be estimated, based upon an average packet size multiplied by the number of nonreceived packets. This value can also be estimated by the control server with the information supplied.
- Number of RTP audio packets received.
- Number of audio bytes received in the RTP payload.
- Estimated interarrival jitter (in milliseconds) Interarrival jitter is an estimate of the statistical variance among the arrival times of RTP packets, which is equivalent to the difference in their relative transit times. Relative transit time is the difference between a packet's RTP timestamp at the sender and the receiver's clock at the time of arrival.

### *ss7nmi debug-level command*

The TAOS unit reports the VoIP call statistics in the output of the `ss7nmi debug-level` command. When the command is entered with the `-s` option, the results displayed include the number of release channel request (RCR) and release channel completed (ACR) messages sent with and without VoIP call statistics, and the number of unknown SS7 VoIP messages. In the following example, new statistics reported for IPDC VoIP calls are shown in bold type:

```
admin> ss7nmi -s
SS7 NAS Messaging Interface (NMI) statistics:
 Initialized successfully: Yes
 Total number of internal errors: 0
 Level of diagnostics: 0
Signaling Layer:
 Current link state: STARTING
 Last generated transaction ID: 1
 Timer T305 (RST1): 1000 ticks - idle
 Number of protocol version errors: 0
 Number of 'message reject' received: 0
 Number of bad packets received: 0
 Number of unknown messages: 0
 Number of unknown SS7Voip messages: 0
 Number of resource conflicts: 0
 Number of release race conditions: 0
 Number of RCR with stats sent: 0
 Number of RCR without stats sent: 0
 Number of ACR with stats sent: 0
 Number of ACR without stats sent: 0
Data Transport Layer:
 Number of link fail-overs: 0
 Number of persistent errors: 0
 Last error: No Error
 Last error timestamp: [01/01/1990 00:00:00]
```

## Statistics and error reporting on SS7 connections

The `ss7asg -s` command provides detailed interface information about statistics and error conditions on SS7 connections. The output differs depending on whether errors are detected.

**Note:** The `ss7asg -r` command resets all the signaling layer statistics to 0 and updates the timestamp to the time the counters were reset.

### *Command output when no errors are detected*

The following sample output indicates that no errors were detected in SS7 connections:

```
admin> ss7asg -s
SS7 Signaling Gateway interface statistics:
 Initialized successfully: Yes
 Interface state: Enabled/Down
 Diagnostic level: 0
Signaling Layer:
 Number of SETUP requests from: L2: 0 CC: 0
```

```

Number of CONNECT to ASG: 0
Number of CONNECT_ACK from ASG: 0
Number of SETUP rejected from: L3: 0 CC: 0
Number of DISCONNECT requests from: L2: 0 CC: 0
Number of REGISTRATION to ASG: 0
Number of REGISTRATION_ACK from ASG: 0
Number of DL_REL_IND from L2: 0
Number of DL_EST_IND from L2: 0
Number of T303 expiry events: 0
Number of T305 expiry events: 0
Number of T308 expiry events: 0
Last L3 counters reset timestamp: [02/08/1999 18:47:41]

Data Transport Layer:
Number of link fail-overs: 0
Number of persistent errors: 161
Last error: Persistent Error
Last error status change timestamp: [02/08/1999 18:47:41]

```

When the command reports no errors, the output contains the following fields:

| Output field                   | Description                                                                                                                                                                                                                                                                                                                                                                                                                |
|--------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Initialized successfully       | Indicates whether the SS7 layer between the TAOS unit and the signaling gateway has been successfully initialized.                                                                                                                                                                                                                                                                                                         |
| Interface state                | State of the SS7 interface. A value of Enabled/Up indicates that the Enabled parameter in the SS7-Gateway profile is set to yes. A value of Enabled/Down indicates that the Enabled parameter in the SS7-Gateway profile is set to yes, but the TCP link to the signaling gateway is down. A value of Disabled indicates that the Enabled parameter in the SS7-Gateway profile is set to no.                               |
| Diagnostic level               | The diagnostic level as specified with the <code>-t</code> option. Values can be one of the following: <ul style="list-style-type: none"> <li>• 0: Disable diagnostic output.</li> <li>• 1: Show errors only.</li> <li>• 2: Trace L3 events and states.</li> <li>• 3: Trace Call Control events.</li> <li>• 4: Show all task events.</li> <li>• 5: Dump L3 packets.</li> <li>• 6: Dump Call Control primitives.</li> </ul> |
| Number of SETUP requests from: | L2: Number of setup requests from the signaling gateway (SS7 network) or from incoming calls.<br><br>CC: Number of times the TAOS unit tried to make an outgoing call to the signaling gateway (the SS7 network). Note that outgoing calls are not currently supported.                                                                                                                                                    |

| Output field                        | Description                                                                                                                                                                                                                                                                                                                                                |
|-------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Number of CONNECT to ASG            | Total number of active connections to the signaling gateway since it was last reset.                                                                                                                                                                                                                                                                       |
| Number of CONNECT_ACK from ASG      | Number of connection acknowledgements the TAOS unit has received from the signaling gateway.                                                                                                                                                                                                                                                               |
| Number of SETUP rejected from:      | Number of setup requests rejected by layer 3 and the signaling gateway call control. <ul style="list-style-type: none"><li>• Setups rejected by L3 indicate a packet decode error on the incoming setup request.</li><li>• Setups rejected by CC can mean that no route or resource exists, or that authentication failed for the incoming call.</li></ul> |
| Number of DISCONNECT requests from: | Number of disconnection requests from layer 2 and the signaling gateway call control. <ul style="list-style-type: none"><li>• Disconnection requests from layer 2 are initiated by the signaling gateway.</li><li>• Disconnection requests from CC are initiated by the TAOS unit.</li></ul>                                                               |
| Number of REGISTRATION to ASG       | Number of registration requests the TAOS unit has sent to the signaling gateway.                                                                                                                                                                                                                                                                           |
| Number of REGISTRATION_ACK from ASG | Number of registration acknowledgments the TAOS unit has received from the signaling gateway.                                                                                                                                                                                                                                                              |
| Number of DL_REL_IND from L2        | Number of Data Link Release Indication messages received from layer 2. Layer 2 sends these messages to layer 3 to inform it about the status of the link. Data Link Release Indication messages mean that the link between the TAOS unit and the signaling gateway is down and communication is not possible.                                              |
| Number of DL_EST_IND from L2        | Number of Data Link Establish Indication messages received from layer 2.<br><br>Layer 2 sends these messages to layer 3 to inform it about the status of the link. Data Link Establish Indication messages mean that the link between the TAOS unit and the signaling gateway has been reestablished and communication is possible.                        |
| Number of T303 expiry events        | Number of times the T303 timer expired.                                                                                                                                                                                                                                                                                                                    |
| Number of T305 expiry events        | Number of times the T305 timer expired.                                                                                                                                                                                                                                                                                                                    |
| Number of T308 expiry events        | Number of times the T308 timer expired.                                                                                                                                                                                                                                                                                                                    |
| Last L3 counters reset timestamp    | Time the signaling layer timers were last reset using the <code>ss7asg -r</code> command.                                                                                                                                                                                                                                                                  |

| Output field                       | Description                                                                                                                                                                                                                                                                                                         |
|------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Number of link fail-overs          | In a dual LAN configuration, the number of times the TAOS unit switched from one TCP/IP messaging link to another due to the failure of the link.                                                                                                                                                                   |
| Number of persistent errors        | Number of times the TAOS unit tried to reestablish a layer 2 link.                                                                                                                                                                                                                                                  |
| Last error                         | Type of last error. Possible values are: <ul style="list-style-type: none"> <li>• No Error: L2 is operating normally.</li> <li>• Link Loss: Link down.</li> <li>• Persistent Error: Link down.</li> <li>• Link Shutdown: Link disabled.</li> <li>• Link Fail-over: Switched to secondary LAN connection.</li> </ul> |
| Last error status change timestamp | Time the last error occurred.                                                                                                                                                                                                                                                                                       |

### *Command output showing errors*

The following sample indicates that errors were detected in SS7 connections:

```
admin> ss7asg -s
SS7 Signaling Gateway interface statistics:
 Initialized successfully: Yes
 Interface state: Enabled/Down
 Diagnostic level: 0

Errors:
 Number of memory allocation failures: 0
 Number of errors in profile operations: 0
 Number of invalid memory pointers: 0
 Number of internal errors: 8

Initialization Errors:
 Number of errors in initialization: 8
 Memory pools: 0
 Mailboxes: 0

Signaling Layer:
 Number of SETUP requests from: L2: 0 CC: 0
 Number of CONNECT to ASG: 0
 Number of CONNECT_ACK from ASG: 0
 Number of SETUP rejected from: L3: 0 CC: 0
 Number of DISCONNECT requests from: L2: 0 CC: 0
 Number of REGISTRATION to ASG: 0
 Number of REGISTRATION_ACK from ASG: 0
 Number of DL_REL_IND from L2: 0
 Number of DL_EST_IND from L2: 0
 Number of T303 expiry events: 0
 Number of T305 expiry events: 0
```



```

Number of T308 expiry events: 0
Last L3 counters reset timestamp: [02/16/1999 10:33:31]

Data Transport Layer:
Number of link fail-overs: 0
Number of persistent errors: 0
Last error: No Error
Last error status change timestamp: [01/01/1990 00:00:00]

```

When errors are detected, the command output displays the fields explained in the previous section plus the following additional information:

| <b>Output field</b>                    | <b>Description</b>                                                                                                                                                                                          |
|----------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Number of memory allocation failures   | Number of times the TAOS unit could not allocate memory for packets traveling between call control and layer 3.<br><br>These errors might occur if the TAOS unit does not have a 32-MB DRAM card installed. |
| Number of errors in profile operations | Number of times the TAOS unit could not register the SS7-Gateway profile or read or update a T1 profile.                                                                                                    |
| Number of invalid memory pointers      | Number of empty packets received by IPDC layer 3. Used for IPDC only.                                                                                                                                       |
| Number of internal errors              | Number of internal errors.                                                                                                                                                                                  |
| Number of errors in initialization     | Number of errors that occurred during the initialization of the SS7 ASG interface.                                                                                                                          |
| Memory pools                           | Number of buffer pool allocations that failed.                                                                                                                                                              |
| Mailboxes                              | Number of failures that occurred during the creation or operation of the mailboxes used for interlayer messaging.                                                                                           |

## ***Cause codes for SS7 ASGCP calls to the TAOS unit***

The TAOS unit reports cause codes to the signaling gateway via ASGCP when it initiates a call clearing. The following ASGCP messages carry cause code information.

- Disconnect
- Release
- Release Complete
- Restart Acknowledgement (cause optional)
- Status

The TAOS unit currently reports the cause codes defined by ITU-T Recommendation Q.850. For definitions of the individual cause values, refer to Q.850. Note the following:

- The TAOS unit reports Normal call clearing (16) if a TAOS unit modem times out on a modem call.

- The TAOS unit reports User busy (17) if it cannot find a route, or if no resource is available for the call.

## **SS7 IPDC support for call ID and disconnect cause codes**

The TAOS unit reports a globally unique call identifier to call-logging servers for SS7 data or VoIP calls. This feature enables the NavisAccess software to associate call statistics information generated by the signaling gateway and by the TAOS unit.

A similar mechanism is supported in the H.323 VoIP context, where a well-defined globally unique call ID is set by the originating endpoint. This call ID is used to associate remote access server (RAS) signaling with the modified Q.931 call control signaling used in H.225.0 call setup. In an H.323 VoIP environment, the TAOS unit reports the call ID to call-logging servers when a call is connected, maintained, and terminated. (H.323, Q.931, and H.225.0 are ITU-T recommendations for voice communication over networks.) For more information about H.323 VoIP see the *MultiVoice for MAX TNT Configuration Guide*.

To support this functionality in the SS7 IPDC context, the following changes were made:

- IPDC now generates a globally unique call ID for SS7 VoIP and data calls.
- IPDC now includes the globally unique call ID in IPDC messages.
- The TAOS unit now reports the call ID to call-logging servers.

### *IPDC generation of a globally unique call ID*

IPDC uses the same definition and algorithm for generating a globally unique call Identifier as H.225.0. The ID consists of a record of 16 octets. For details, refer to H.225.0, Version 2, pages 44 to 47. The signaling gateway creates the call ID in the following cases:

- Request inbound call setup (RCSI) message
- Request pass-through call setup for TDM connection between two channels (RCST) message
- Request packet pass-through call (RCCP) message

The TAOS unit creates a call ID for a request outbound call setup (RCSO) message. Note that the TAOS unit does not currently report the call ID of outbound calls to call-logging servers.

### *Global-Call-ID parameter*

The Global-Call-ID parameter in the Call-Info profile reports the global call ID and is shown with a sample setting in the following example:

```
[in CALL-INFO/{ 3 }]
mbid* = { 3 }
call-service = switched
called-number-type = 2
nailed-up-group = 1
call-by-call = 0
phone-number = " "
transit-number = " "
billing-number = " "
switched-call-type = 67
```

```
ft1-caller = 0
calling-number = { "" unknown unknown unspecified unspecified }
force-56kbps = 0
redirect-number = ""
call-direction = 0
global-call-id = 03040506-0102-0900-0807-010203040506
```

### *Start and Stop records*

The Ascend\_Global\_Call\_Id attribute in the Start and Stop records for SS7 VoIP and data calls is for call-logging only, not RADIUS, and is reported only when the global call ID is available.

The TAOS unit sends Stop records for SS7 calls that are cleared or rejected at the SS7 IPDC layer. Those calls do not have Start records, because they are never routed to host cards.

### *Disconnect cause codes*

The following set of disconnect cause codes reports the cause of termination for calls that are cleared or rejected at the SS7 IPDC layer. These codes are based on the cause codes defined by ITU-T Recommendation Q.850, *Usage of Cause and Location in the Digital Subscriber Signaling System No. 1 and the Signaling System No. 7 ISDN User Part*. This group of cause codes begins at offset 800.

| Event                          | Code | Q.850 Definition                           |
|--------------------------------|------|--------------------------------------------|
| DIS_Q850_UNASSIGNED_NUMBER     | 801  | Unallocated (unassigned) number            |
| DIS_Q850_NO_ROUTE              | 802  | No route to specified transit network      |
| DIS_Q850_NO_ROUTE_TO_DEST      | 803  | No route to destination                    |
| DIS_Q850_CHANNEL_UNACCEPTABLE  | 806  | Channel unacceptable                       |
| DIS_Q850_NORMAL_CLEARING       | 816  | Normal call clearing                       |
| DIS_Q850_USER_BUSY             | 817  | User busy                                  |
| DIS_Q850_NO_USER_RESPONDING    | 818  | No user responding                         |
| DIS_Q850_USER_ALERT_NO_ANSWER  | 819  | No answer from user (user alerted)         |
| DIS_Q850_CALL_REJECTED         | 821  | Call rejected                              |
| DIS_Q850_NUMBER_CHANGED        | 822  | Number changed                             |
| DIS_Q850_DEST_OUT_OF_ORDER     | 827  | Destination out of order                   |
| DIS_Q850_INVALID_NUMBER_FORMAT | 828  | Invalid number format (address incomplete) |
| DIS_Q850_FACILITY_REJECTED     | 829  | Facility rejected                          |
| DIS_Q850_RESP_TO_STAT_ENQ      | 830  | Response to STATUS ENQUIRY                 |
| DIS_Q850_UNSPECIFIED_CAUSE     | 831  | Unspecified normal event                   |
| DIS_Q850_NO_CIRCUIT_AVAILABLE  | 834  | No circuit or channel available            |
| DIS_Q850_NETWORK_OUT_OF_ORDER  | 838  | Network out of order                       |

## Signaling System 7 (SS7)

### Cause codes for SS7 ASGCP calls to the TAOS unit

---

| Event                            | Code | Q.850 Definition                                                                       |
|----------------------------------|------|----------------------------------------------------------------------------------------|
| DIS_Q850_TEMPORARY_FAILURE       | 841  | Temporary failure                                                                      |
| DIS_Q850_NETWORK_CONGESTION      | 842  | Switching equipment congestion                                                         |
| DIS_Q850_ACCESS_INFO_DISCARDED   | 843  | Access information discarded                                                           |
| DIS_Q850_REQ_CHANNEL_NOT_AVAIL   | 844  | Requested circuit or channel not available                                             |
| DIS_Q850_PRE_EMPTED              | 845  | Call preempted                                                                         |
| DIS_Q850_RESOURCE_NOT_AVAIL      | 847  | Resource unavailable                                                                   |
| DIS_Q850_FACILITY_NOT_SUBSCRIBED | 850  | Requested facility not subscribed                                                      |
| DIS_Q850_OUTGOING_CALL_BARRED    | 852  | Outgoing calls barred within the CUG                                                   |
| DIS_Q850_INCOMING_CALL_BARRED    | 854  | Incoming calls barred within the CUG                                                   |
| DIS_Q850_BEAR_CAP_NOT_AVAIL      | 858  | Bearer capability not presently available                                              |
| DIS_Q850_SERVICE_NOT_AVAIL       | 863  | Service or option not available, unspecified                                           |
| DIS_Q850_CAP_NOT_IMPLEMENTED     | 865  | Bearer capability not implemented                                                      |
| DIS_Q850_CHAN_NOT_IMPLEMENTED    | 866  | Channel type not implemented                                                           |
| DIS_Q850_FACILITY_NOT_IMPLEMENT  | 869  | Requested facility not implemented                                                     |
| DIS_Q850_INVALID_CALL_REF        | 881  | Invalid call reference value                                                           |
| DIS_Q850_CHAN_DOES_NOT_EXIST     | 882  | Identified channel does not exist                                                      |
| DIS_Q850_INCOMPATIBLE_DEST       | 888  | Incompatible destination                                                               |
| DIS_Q850_MANDATORY_IE_MISSING    | 896  | Mandatory information element missing                                                  |
| DIS_Q850_NONEXISTENT_MSG         | 897  | Message type nonexistent or not implemented                                            |
| DIS_Q850_WRONG_MESSAGE           | 898  | Message not compatible with call state, or message type nonexistent or not implemented |
| DIS_Q850_NONEXISTENT_IE          | 899  | Information element or parameter nonexistent or not implemented                        |
| DIS_Q850_INVALID_ELEM_CONTENTS   | 900  | Invalid information element contents                                                   |
| DIS_Q850_WRONG_MSG_FOR_STAT      | 901  | Message not compatible with call state                                                 |
| DIS_Q850_TIMER_EXPIRY            | 902  | Recovery on timer expiration                                                           |
| DIS_Q850_MANDATORY_IE_LEN_ERR    | 903  | Parameter that was nonexistent or not implemented was passed on                        |

| Event                        | Code | Q.850 Definition                                  |
|------------------------------|------|---------------------------------------------------|
| DIS_Q850_PROTOCOL_ERROR      | 911  | Message with unrecognized parameter was discarded |
| DIS_Q850_INTERWORKING_UNSPEC | 927  | Unspecified internetworking event                 |

## SNMP support for SS7

The SS7 MIB (`mgstat.mib`) is implemented as a branch object with the main object, `mgGroup`, linked into the Ascend enterprise MIB. For definitions and descriptions of objects, see the `mgstat.mib` file distributed with TAOS 8.0.2 software.

An SNMP trap is supported for reporting the status of the link between SS7 media gateways and the TAOS unit. The trap can be configured when an SS7 license is enabled. For a trap to be generated when the trap condition occurs, SNMP traps must be enabled and the setting for the trap condition must be enabled. For details about enabling traps, see the *APX 8000/MAX TNT/DSLNT Administration Guide*.

The following trap has been added to the Ascend enterprise traps:

```
megacoLinkStatusTrap TRAP-TYPE
 ENTERPRISE ascend
 VARIABLES { mgLinkName, mgOperStatus }
 DESCRIPTION "This trap indicates that operational status
 of a media gateway control link has changed."
 ::= 42
```

Following is the relevant parameter in the Trap profile, shown with its default value, for enabling the trap:

```
[in TRAP/""]
megaco-link-status-enabled = no
```

| Parameter                  | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|----------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Megaco-Link-Status-Enabled | <p>Enable/disable trap generation of communication link status between the SS7 media gateway and the TAOS unit. This trap indicates that operational status of a media gateway control link has changed from any state to the Up state or from Up state to any other state. Changes to this parameter become effective when you write the Trap profile.</p> <p>The trap contains the name of the link, which is currently always reported as <code>default</code>, and the new operational status.</p> |

For example, the following commands enable the SS7 link-state trap:

```
admin> read trap example
TRAP/example read

admin> set megaco-link-status-enabled = yes

admin> write
TRAP/example written
```



# Configuring Call Routing

# 19

|                                                        |       |
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The TAOS unit uses a set of call-routing algorithms to route inbound and outbound calls to devices that handle the appropriate call type. For example, the unit routes an inbound voice-service call to a modem and an inbound digital-service call to an HDLC channel.

**Note:** For the APX 8000, slots are divided into four quadrants of 10 slots each to increase the efficiency of routing among shelves. To maintain maximum call capacity and efficient use of resources, you must balance the network-side and host-side resources in each quadrant, as described in the *APX 8000 Hardware Installation Guide*.

The TAOS unit creates Call-Route profiles that specify generalized routes to its devices. The generalized routes, which typically route on the basis of call type, make up a default call-routing database. The default database is just a starting point, representing the unit's best guesses for appropriate call handling. You can create additional Call-Route profiles that override or complement the default call routes.

**Note:** The system does not route calls to a device that has no applicable entry in the call-routing database, so be careful not to delete default entries without providing a replacement.

## ***Network, host, and dual-purpose devices***

Slot cards that are used to establish and maintain the physical connection for a call are *network cards*. Network cards do not support protocol stacks such as PPP or Frame Relay. Instead, they rely on another card, such as a Hybrid Access card, to remove link encapsulation and process the call's protocol information and thereby terminate the call in the system.

Series56 II and Series III cards, MultiDSP cards, Hybrid Access (HDLC) cards, modem cards, and some other cards that terminate inbound calls are referred to as *host cards*. An individual channel or modem on a host card is referred to as a *host device*.

Some slot cards act in a *dual* capacity, performing both network and host functions. Some dual-purpose cards provide their own HDLC resources. For example, T1 and E1 FrameLine cards have HDLC channels integrated into the card and support Frame Relay protocols. Other cards establish the physical connection and terminate calls. Calls on these cards do not require additional HDLC processing.

The following table shows a representative listing of cards in each category:

| Network slot cards | Host slot cards            | Dual-purpose slot cards |
|--------------------|----------------------------|-------------------------|
| T1                 | Series56 II Digital Modem  | Unchannelized DS3       |
| T3                 | Series56 III Digital Modem | T1 FrameLine            |
| E1                 | Hybrid Access              | E1 FrameLine            |
|                    | MultiDSP                   |                         |

## ***Understanding the call-routing database***

The Callroute command displays entries in the call-routing database. You can display entries for network, host, or dual-purpose devices, or list entries by device. For example, the Callroute command with the -ad flag displays database entries for dual-purpose devices:

```
admin> callroute -ad
 device # source type tg sa phone
1:16:01/0 0 0:00:00/0 digital-call-type 0 0
2:14:01/0 0 0:00:00/0 digital-call-type 0 0
1:15:01/0 0 0:00:00/0 any-call-type 0 0
2:04:01/0 0 0:00:00/0 any-call-type 0 0
2:04:02/0 0 0:00:00/0 any-call-type 0 0
2:04:03/0 0 0:00:00/0 any-call-type 0 0
2:04:04/0 0 0:00:00/0 any-call-type 0 0
```

Table 19-1 describes the information contained in each field.

*Table 19-1. Fields in a call-routing database entry*

| Field  | Contains                                                                                                                                                                                                                                                                                                      |
|--------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Device | The address of a device installed in the system to which calls matching the entry will be routed. The address has the following format:<br><br><i>shelf:slot:line/channel</i><br><br>For example, shelf 1, slot 2, line 1, channel 24 is addressed as 1:02:01/24.                                             |
| #      | The call-routing entry number. For default entries (entries created by the system for a device) the value is always zero. For entries created by Call-Route profiles, the number must be unique for each entry that has the same Device address. For details, see “Call-Route profile settings” on page 19-5. |



*Table 19-1. Fields in a call-routing database entry (continued)*

| Field  | Contains                                                                                                                                                                                                                                 |
|--------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Source | The address of a device that receives calls. The address has the same format as the Device field.                                                                                                                                        |
| Type   | The call-routing type. For default entries (entries created by the system) the value depends on the type of installed device. For entries created by user-specified Call-Route profiles, see “Call-Route profile settings” on page 19-5. |
| TG     | A trunk group number. The default is zero.                                                                                                                                                                                               |
| SA     | An ISDN subaddress number. The default is zero.                                                                                                                                                                                          |
| Phone  | A telephone number.                                                                                                                                                                                                                      |

## How call routes affect device usage

The system sorts its call-routing database entries after a reset. During active use, the system applies operational criteria, such as availability and frequency of use, to the order in which it uses devices.

By default, entries for modem devices are sorted to load-balance calls across the modem cards in a quadrant, and entries for HDLC channels are sorted to group the channels of a multilink call on a single card when possible. For both types of calls, cards in lower-numbered slots precede those in higher-numbered slots.

Entries for trunk lines are initially sorted in the order in which the lines are installed in the system, with lines in lower-numbered slots preceding those in higher-numbered slots.

## Modem usage and database sort order

The default sort order for modems is determined by the following parameter, shown with its default settings:

```
[in SYSTEM]
call-routing-sort-method = item-first
```

In a device address, the item number represents a particular device on a slot card. An item number of 0 (zero) denotes the whole slot, and the numbering begins with the leftmost device on the card. The Item-First sort method means the system sorts the components of device addresses in the following order:

*item:shelf:slot:logical:item*

For example, if two modem cards are supported, one in shelf 1, slot 2 and the other in shelf 1, slot 4, the initial modem usage order looks like this:

```
1:1:2
1:1:4
2:1:2
2:1:4
3:1:2
```

```
3:1:4
...
```

The system uses the first modem on the lower-numbered card, then the first modem on the next card. It then uses the second modem on each of the cards, and so forth. This sort order causes the channels of different cards to be interspersed, resulting in load balancing across all cards that match a call's parameters, even after a system reset.

## HDLC channel usage and database sort order

The default sort order for HDLC channels is determined by the following parameter, shown with its default settings:

```
[in SYSTEM]
digital-call-routing-sort-method = slot-first
```

The Slot-First sort method means the system sorts the components of device addresses in the following order:

*shelf:slot:item:logical item*

For example, if two Hybrid Access (HDLC) cards are supported, one in shelf 1, slot 14 and the other in shelf 1, slot 16, the initial HDLC channel usage order looks like this:

```
1:14:1
1:14:2
1:14:3
...
1:14:96
1:16:1
1:16:2
1:16:3
...
1:16:96
```

With the Slot-First sort order, the system starts with the card in the lowest-numbered slot, and moves on to the next slot card only when all of the devices on the first card have been used. Once a device has been used, it is placed at the end of the sorted list.

Slot-First is the default for digital calls because performance is improved when all channels of a multilink call are on the same card. For example, suppose the system establishes the base channel of an MP+ call on a Hybrid Access card in slot 4. When the connection requires additional bandwidth, the Slot-First sort order gives the TAOS unit a good chance of adding the new channels on slot 4 as well, which results in greater efficiency in handling the call.

However, suppose slot 3 contains a Series56 II card (which provides 48 HDLC channels but does not support Frame Relay connections), slot 4 contains a Hybrid Access card, and the system supports Frame Relay datalinks on T1. Because T1 is a network device, it requires HDLC processing by a host card. In this case, the Slot-First algorithm causes the system to attempt to use each one of the Series56 II card's HDLC channels before moving on to the Hybrid Access card. This behavior can result in up to 48 call rejects before the connection is successfully established. (No system messages are reported during the interval.) To prevent this situation, you can either remove the default Call-Route profile that enables the Series56 II card to handle digital calls, or install the Series56 II card in a higher-numbered slot than the

Hybrid Access card. See “Dedicating Series56 cards to modem processing” on page 19-9 for related information.

## Trunk line usage and sort order

Trunk lines are sorted in the order in which they are installed in the system, with lines in lower-numbered slots preceding those in higher-numbered slots. For example, if more than one E1 card is installed, the system uses the card in the lower-numbered slot first, and begins to use the second E1 card only when the first one is fully utilized.

**Note:** This method of trunk line usage introduces a requirement for explicit Call-Route profiles when more than one trunk group is configured for the system. For details, see “Outbound call routing by trunk group” on page 19-6.

## Working with Call-Route profiles

Administrators create Call-Route profiles to control device usage. While there are many reasons for creating Call-Route profiles, explicit Call-Route profiles are strongly recommended or required in the following cases:

- When trunk groups are in use
- To bundle Multilink PPP calls efficiently
- To reserve HDLC devices for multilink Frame Relay connections

## Call-Route profile settings

Following are the parameters in a Call-Route profile, shown with default settings:

```
[in CALL-ROUTE/{ { any-shelf any-slot 0 } 0 } 0 }]
index* = { { { any-shelf any-slot 0 } 0 } 0 }
trunk-group = 0
phone-number = "
preferred-source = { { any-shelf any-slot 0 } 0 }
call-route-type = any-call-type
```

| Parameter   | Specifies                                                                                                                                                                                                                                                                                                                                                                                            |
|-------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Index       | Destination of the call route. A device address is followed by a call-routing database entry number (starting with zero) in the following format:<br><br><code>{ { shelf slot item } logical-item } entry }</code><br><br>If you create more than one call route for the same destination, the entry numbers must be unique for each Call-Route profile. Entry numbers do not have to be sequential. |
| Trunk-Group | Trunk group number. Enables the system to route calls to the specified destination on the basis of trunk-group information provided by a call.                                                                                                                                                                                                                                                       |

| Parameter        | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                              |
|------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Phone-Number     | Telephone number assigned to TAOS unit network lines. For lines that use ISDN service, the telephone number can contain a subaddress, as described in “Second pass: ISDN subaddresses” on page 19-12.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                  |
| Preferred-Source | Source of a call. The address of a device in the system.                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                               |
| Call-Route-Type  | Type of call to be routed to the device. Valid settings include the following: <ul style="list-style-type: none"><li>• Any-Call-Type (any of the types listed below).</li><li>• Voice-Call-Type (voice bearer calls, which do not include 3.1KHz audio call types or VoIP calls).</li><li>• Digital-Call-Type (general digital calls, including 3.1KHz audio bearer channel calls, routed to a host device).</li><li>• Trunk-Call-Type (digital calls sent to a trunk device). The Trunk-Call-Type setting is used for outbound call routing for trunk calls and trunk-to-trunk switching.</li><li>• PHS-Call-Type (Personal Handyphone System calls).</li><li>• VoIP-Call-Type (Voice-over-IP calls).</li><li>• V110-Call-Type (digital calls recognized as containing V.110 rate-adapted bearer channels).</li></ul> |

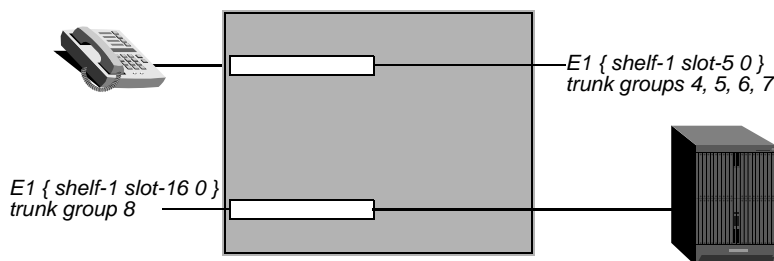
## Outbound call routing by trunk group

If no explicit call routes are defined, the TAOS unit always routes a call to *the first* entry in its database that matches the call’s parameters and is in the same quadrant. Because default entries do not include trunk group specifications, this behavior can cause a call to fail when multiple T1 or E1 cards are installed and each card supports different trunk groups. The system always tries the first installed card, and does not proceed to the second card without an explicit call route instructing it to do so.

**Note:** When configuring trunk groups for the purpose of outbound call routing, you must specify the trunk groups both in the Call-Route profiles and in each channel subprofile of each network line profile.

In Figure 19-1, the system has two E1 cards installed, one in slot 5 and one in slot 16. The channel subprofiles of the E1 lines on the first card specify trunk groups 4, 5, 6, and 7. All channels on all of the E1 lines on the second card specify trunk group 8.

*Figure 19-1. Trunk group 8 connecting to a TAOS unit*



Without an explicit call route for trunk group 8, the system always tries the first E1 card, finds that it does not use trunk group 8, and then drops the call.

The following commands create an explicit call route for trunk group 8:

```
admin> new call-route { { { 1 16 0 } 0 } 0 }
CALL-ROUTE/{ { { shelf-1 slot-16 0 } 0 } 0 } read

admin> set trunk-group = 8

admin> write
CALL-ROUTE/{ { { shelf-1 slot-16 0 } 0 } 0 } written
```

This profile creates a call-routing database entry such as the following:

| device    | # | source    | type            | tg | sa | phone |
|-----------|---|-----------|-----------------|----|----|-------|
| 1:16:01/1 | 0 | 0:00:00/0 | trunk-call-type | 8  | 0  |       |

When the system attempts to bring up a call to the remote TAOS unit on trunk group 8, it matches the trunk-group field in its database and directs the call to the E1 card in slot 16.

## Multilink Frame Relay requirements with Hybrid Access

To implement a Multilink Frame Relay (MFR) bundle using a T1, E1, or T3 card with a Hybrid Access card, you must ensure that the aggregate bandwidth is bound to the channels of a single Hybrid Access card. So, if more than one Hybrid Access card is installed, you must define Call-Route profiles to map the bandwidth of the MFR bundle to the same Hybrid Access card.

**Note:** Because one Hybrid Access card can provide 186 channels (31 x 6) for MFR, one Hybrid Access card can support up to six Call-Route profiles binding its channels to up to six back-to-back E1 ports. This places a six-line limitation on the size of the MFR bundle when you are using an Hybrid Access card.

### *Example with two E1 lines in an MFR bundle*

In the following example, the administrator creates two Call-Route profiles for the Hybrid Access card in slot 3, with each profile binding 31 HDLC channels to a single E1 line on the card in slot 2. The default Call-Route profile for the Hybrid Access card must be left unmodified, or can be deleted, but should not be modified to specify an explicit route.

For example, the following commands create a Call-Route profile for the Hybrid Access card in slot 3 and set the preferred source to the first E1 interface in slot 2:

```
admin> new call-route { { { shelf-1 slot-3 0 } 0 } 1 }
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 1 } read

admin> set preferred-source = { { 1 2 1 } 0 }

admin> list
[in CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 1 } (new) (changed)]
index* = { { { shelf-1 slot-3 0 } 0 } 1 }
trunk-group = 0
phone-number = ""
preferred-source = { { shelf-1 slot-2 1 } 0 }
call-route-type = digital-call-type

admin> write
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 1 } written
```

The next set of commands creates another Call-Route profile for the Hybrid Access card and sets the preferred source to the second E1 interface in slot 2:

```
admin> new call-route { { { shelf-1 slot-3 0 } 0 } 2 }
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 1 } read

admin> set preferred-source = { { 1 2 2 } 0 }

admin> write
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 2 } written
```

Note that the default Call-Route profile for the Hybrid Access card was not modified. It still specifies a general route for the card as a whole, as shown in the following listing:

```
admin> get call-route { { { shelf-1 slot-3 0 } 0 } 0 }
[in CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 0 }]
index* = { { { shelf-1 slot-3 0 } 0 } 0 }
trunk-group = 0
phone-number = ""
preferred-source = { { any-shelf any-slot 0 } 0 }
call-route-type = digital-call-type
```

### *Example with six E1 lines in an MFR bundle*

If the MFR bundle aggregates enough bandwidth to utilize all of the channels on a Hybrid Access card (up to 186, or six E1 lines), you can create a single Call-Route profile mapping the E1 card to the Hybrid Access card. Only six of the E1 lines are usable for MFR, however.

For example, the following commands modify the default Call-Route profile to specify the E1 card in slot 2 as the preferred source for the card:

```
admin> read call-route { { { shelf-1 slot-3 0 } 0 } 0 }
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 0 } read

admin> set preferred-source = { { 1 16 0 } 0 }

admin> list
[in CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 0 } (changed)]
index* = { { { shelf-1 slot-3 0 } 0 } 0 }
trunk-group = 0
phone-number = ""
preferred-source = { { shelf-1 slot-16 0 } 0 }
call-route-type = digital-call-type

admin> write
CALL-ROUTE/{ { { shelf-1 slot-3 0 } 0 } 0 } written
```

## **Concentrating multilink calls on one Hybrid Access card**

Multilink calls that add channels dynamically might inadvertently use channels distributed across multiple Hybrid Access cards, which causes a performance penalty for those calls. You can use Call-Route profiles to direct the system to route calls received on a particular trunk to a single Hybrid Access card.

In this example, the system has three T1 cards installed in slots 1, 2, and 3. Slot 1 uses trunk groups 4 and 5, slot 2 uses trunk groups 6 and 7, and slot 3 uses trunk groups 8 and 9. The system also has three Hybrid Access cards, in slots 4, 5, and 6.

The following commands create a Call-Route profile to route calls received on trunk group 4 to the Hybrid Access card in slot 4:

```
admin> new call-route { { { 1 4 0 } 0 } 0 }
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 0 } read

admin> set trunk-group = 4

admin> write
CALL-ROUTE/{ { { shelf-1 slot-4 0 } 0 } 0 } written
```

This profile creates a call-routing database entry such as the following:

| device    | # | source    | type              | tg | sa | phone |
|-----------|---|-----------|-------------------|----|----|-------|
| 1:04:01/1 | 0 | 0:00:00/0 | digital-call-type | 4  |    | 0     |

When the system receives an add-channel request from the caller, it searches the call-routing database for available HDLC channels. This entry ensure that when the add-on request is made on trunk group 4, the HDLC channels will reside on the card in slot 4.

## Dedicating Series56 cards to modem processing

The Series56 II and Series56 III cards handle both modem and digital calls. The TAOS unit automatically creates two Call-Route profiles when you first install one of the cards: one profile for voice call type (a modem call) and one for digital calls. For more information, see “Series56 II and III Call-Route profiles” on page 5-4.

If you want the card to answer only modem calls, delete the Digital-Call-Type profile.

## Enabling Series56 cards to handle HDLC processing

If you want Series56 II and Series56 III cards to answer HDLC calls, then no matter where you install the card, you might experience delays as it tries to answer single channel nailed Frame Relay calls.

To reduce such delays, install a Hybrid Access (HDLC) card in a lower-numbered slot than any Series56 II or Series56 III card. This arrangement enables the Hybrid Access card to answer the Frame Relay call first. However, if all the channels in the Hybrid Access card are in use, or have been used before, the TAOS unit looks for the next available channel, which might be an HDLC channel in the Series56 II or Series56 III card.

## ***Another way to route incoming calls (deprecated)***

Many network line profiles provide a parameter for directing incoming calls to a particular host interface. Following are the relevant parameters, shown with default settings:

```
[in T1/{ any-shelf any-slot 0 }:line-interface:channel[1]]
call-route-info = { any-shelf any-slot 0 }

[in E1/{ any-shelf any-slot 0 }:line-interface:channel[1]]
call-route-info = { any-shelf any-slot 0 }
```

| Parameter       | Specifies                                                                                                                                                                                                                                                                                                                                                                                                                                           |
|-----------------|-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Call-Route-Info | Address of an interface to which a call can be routed, using the following format:<br><br><code>{ shelf slot item }</code><br><br>The default value is nonspecific, with zero values in each field. This parameter is deprecated. Use of Call-Route profiles is preferred. However, if you specify both methods, the Call-Route-Info setting takes precedence. For more information, see “Fourth pass: destination device addresses” on page 19-13. |

For example, the following commands specify that calls received on the tenth T1 channel on line 1 of a card installed in shelf 1, slot 22 is always routed to the specified host device:

```
admin> read t1 {1 2 1}
T1/{ shelf-1 slot-22 1 } read

admin> set line channel 10 call-route-info = { 1 7 12 }

admin> write
T1/{ shelf-1 slot-22 1 } written
```

## ***Call routing algorithms***

The call-routing database starts with a list of all possible destinations in the system. During active use, the TAOS unit keeps track of the devices that are currently in use and does not consider those devices as possible destinations for a call. After removing entries for devices that are in use, the system sorts the list of remaining devices in the following order:

- Trunk group number (sorted in descending order; for example, 9–4)
- Subaddress number (sorted in descending order; for example, 9–1)
- Telephone number (empty telephone numbers last)
- Destination device address (zero components sorted after nonzero components)
- Source device address (zero components sorted after nonzero components)
- Routing type (Any-Call-Type last)

After sorting the database in this order, the TAOS unit sorts again, this time placing devices that have been used less frequently ahead of those that have been used more frequently.

## **Localization of call routes within a quadrant**

The TAOS unit uses the same call-routing algorithms as similar devices that have a lower port density, such as the MAX TNT, except that it searches first for available devices within the same quadrant as the source of the call. For example, when a call arrives on a network line, the TAOS unit searches first for host-side devices within the quadrant. If no devices are available within the quadrant, the system searches the other quadrants looking for a free entry. To take advantage of this algorithm, you must balance the number of network and host devices within each quadrant.



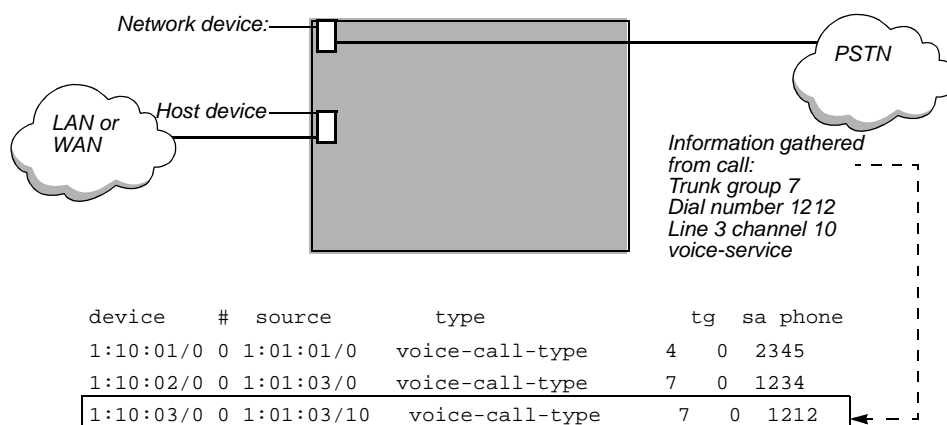
## How the system finds a route

After sorting the call-routing database, the TAOS unit compares the information it has gathered about a call to the values in the database, looking for a match. Values are compared in the sorted order.

After each comparison, profiles that do not contain zero or a matching value in that compared field are dropped from consideration, so after each pass the list is narrowed considerably. The TAOS unit routes the call to the best match, which is the device that matches the greatest number of components without the use of zero fields. If more than one device matches the call's parameters equally, the system routes the call to the first matching entry.

For example, Figure 19-2 shows three database entries for a modem card in slot 6, all of which were created by Call-Route profiles.

Figure 19-2. Matching call information to a database entry



In the first entry, the preferred source is set to the first T1 line in slot 1, and the dial number is 2345. This entry passes the trunk group and subaddress comparisons, because both fields specify 0. It fails on the telephone number comparison and is dropped from the list.

The second entry sets the preferred source to any channel on the third T1 line in slot 1. It specifies trunk group 7 and dial number 1234. This entry passes the trunk group and subaddress comparisons, because the trunk group matches and the subaddress is 0. It fails on the telephone number comparison and is dropped from the list.

The third entry sets the preferred source to channel 10 on the third T1 line in slot 1. It specifies trunk group 7 and dial number 1212. This entry matches the call information, so the call is routed to the third modem in slot 6.

**Note:** If the list of remaining devices becomes empty at any point, the TAOS unit drops the call. Depending on the type of call, the signaling being used, and the configuration of the central office (CO) switch, dropping the call might result in the switch returning a busy indication to the caller. If the caller receives a busy indication on a voice line, the indication originates from the central office switch equipment, not from the TAOS unit.

## Details of how a route is chosen

The system compares call information to its database entries. The entries are accessed in the order in which they are sorted, with devices that are used less frequently preceding those that have been used recently, and the following sort orders:

- Trunk group number (sorted in descending order; for example, 9–4)
- Subaddress number (sorted in descending order; for example, 9–1)
- Telephone number (empty telephone numbers last)
- Destination device address (zero components sorted after nonzero components)
- Source device address (zero components sorted after nonzero components)
- Routing type (Any-Call-Type last)

### *First pass: trunk group number*

The first pass through the database compares trunk-group information gathered from the call to the trunk-group numbers in entries. Entries with a matching trunk-group number or a trunk group of zero remain in the list for the next comparison pass. Entries with a different trunk-group number are dropped for the next pass. For example, if the input trunk group is 9, profiles with a trunk group of 0 or 9 remain in the list.

### *Second pass: ISDN subaddresses*

If an ISDN subaddress is configured on a line, callers must include the number in the dial number. For example, the caller dials 510-555-1212, 3 where 3 is the subaddress number. Specifying a subaddress as part of the telephone number makes the telephone number much more specific. Only calls that specify the subaddress will match this parameter.

If an inbound call contains an ISDN subaddress as part of the telephone number (for example, the 3 in 510-555-1212, 3), the TAOS unit compares that subaddress to the Phone-Number parameters in its call routing database and rejects entries that specify a different subaddress. Only profiles that specify *the same* subaddress as the one presented by the call remain in the list, unless the TAOS unit finds *no* profiles with a matching subaddress. In that case, it keeps profiles with no subaddress specification in the list and uses them in the next comparison pass.

For example, if the input subaddress is 9, only devices that specify a subaddress of 9 in the Phone-Number parameter remain in the list. Profiles with specifications such as the following, for example, remain in the list:

```
phone-number = 9,
phone-number = 9,555-1212
```

If no devices specify the subaddress 9, only devices with no subaddress specification remain in the list. For example:

```
phone-number = 555-1212
phone-number = 777-9898
```

### *Third pass: telephone numbers*

The TAOS unit compares the telephone number on which the call was received to the Phone-Number values in its call-routing database and rejects all entries with nonmatching numbers. To match the telephone number, an entry's telephone number must be smaller than or

equal to the input number, and its digits must match the add-on digits of the input number. For example, suppose the calling switch sent the following number to the TAOS unit:

555-1212

Profiles with the following telephone numbers (for example) would remain in the list:

```
phone-number = 1212
phone-number = 555-1212
phone-number = 12
```

As with subaddress routing, if the TAOS unit finds no matching telephone numbers, it drops the profiles that have other, nonmatching numbers, but retains the profiles that have a null Phone-Number specification.

#### *Fourth pass: destination device addresses*

The next comparison uses destination-address information specified in the Call-Route-Info parameter of the channel configuration, if the network port configuration has an assigned value for that parameter. By default, the Call-Route-Info parameter specifies the system address:

```
{ any-shelf any-slot 0 }
```

If a call comes in on a channel that specifies a host device address instead, the TAOS unit excludes all profiles whose index does not match that address. The TAOS unit uses the most specific address match. For example, if the channel Call-Route-Info address is { 1 5 1 }, the TAOS unit uses the entry for { 1 5 1 }, if one exists. If there is no entry for { 1 5 1 }, uses the entry for { 1 5 0 }. If there is no entry for { 1 5 0 }, it uses the entry for { 1 0 0 }. If it does not find an entry in the call-routing database for { 1 0 0 }, the TAOS unit uses the default call route, which has the system address, { 0 0 0 }.

#### *Fifth pass: source device addresses*

Next, the TAOS unit compares the device address of the line and channel on which the call was received to the preferred source addresses in its call-routing database, and rejects all profiles with nonmatching preferred source addresses. The default preferred-source address { 0 0 0 } matches all calls.

#### *Last pass: comparison routing type*

For all profiles that remain as possible route destinations after the preceding comparison passes, the TAOS unit compares the type of the incoming call to the Call-Route-Type setting in its call-routing database.

Call type is information that the TAOS unit can detect about any call it receives. The information might indicate the bearer capability of the call, or it might be related to characteristics of the calling device. For example, analog modems place voice-service calls. ISDN devices generally place data-service (digital) calls, but in some cases can place data-over-voice calls.

The TAOS unit excludes all profiles whose routing type does not match the characteristics of the calling device or the bearer capability of the call. For example, if the incoming call uses

voice service, all profiles that specify Digital-Call-Type are removed from consideration. Only profiles that specify Voice-Call-Type or Any-Call-Type remain in the list.

**Note:** For T1 lines that use inband signaling, bearer-capability is not known. The TAOS unit treats all calls that terminate on a T1 and use inband signaling as digital calls unless the T1 profile sets the default call type to voice.

# Provisioning the Switch

|                                                   |     |
|---------------------------------------------------|-----|
| Provisioning the switch for T1 access .....       | A-1 |
| What you need from your T1 service provider ..... | A-2 |
| What you need from your E1 service provider ..... | A-2 |

This appendix provides the information necessary for properly provisioning a switch for T1/EI or T1/E1 PRI access to the WAN.

## *Provisioning the switch for T1 access*

If you use an inband signaling line, the T1 circuit at the Point-of-Presence (POP) must support the translations listed in Table A-1 for compatibility with the DSLTNT.

*Table A-1. T1 access provisioning information*

| Translation                                       | Optional or required                                |
|---------------------------------------------------|-----------------------------------------------------|
| Two-state DTMF (Dual-Tone Multifrequency) dialing | Required for outdial.                               |
| Outgoing wink start                               | Required for outdial.                               |
| Incoming immediate seizure                        | Optional for a switch.                              |
| Incoming wink start                               | Optional for a switch.                              |
| Incoming digits suppressed                        | Required.                                           |
| Answer supervision                                | Required.                                           |
| Switched data                                     | Required.<br>No voice/digital loss plan is allowed. |

Four-state A-bit signaling, four-state B-bit signaling, and pulse dialing are not supported.

## ***What you need from your T1 service provider***

Request the following information about your T1 interface from your WAN provider:

- Type of signaling (inband or ISDN D channel)
- Type of line encoding (B8ZS or AMI)
- Type of framing (ESF or D4)
- Each telephone number assigned to the line, on a channel-by-channel or service-by-service basis
- Number of nailed-up channels, if any
- Number of unused channels, if any
- Types of call-by-call services (also called NSF identifiers) on the switched channels
- Type of line provisioning (B channel, H0 channel, H11 channel, or multirate)
- D-channel assignment
- NFAS ID number (if the T1 PRI line is provisioned for NFAS)

Also, keep in mind the following points:

- In general, ESF framing and B8ZS line encoding are both recommended for T1 applications. In addition, channel 24 must be the D channel, except for applications using non-facility associated signaling (NFAS).
- Applications that require NFAS must be connected to an AT&T or Northern Telecom switch provisioned with NFAS.
- The TAOS unit can receive multichannel calls using MP encapsulation only if all channels of the call share a common telephone number (namely, a hunt group). You can request that your service provider supply you with a hunt group.

## ***What you need from your E1 service provider***

Request the following information about your E1 interface from your WAN provider:

- The telephone numbers assigned to your E1 interface, channel by channel
- Nailed-up channels (also called private WAN), if any
- Unused channels, if any
- Switch type (or emulation)—DPNSS only
- Configuration for switch layers 2 and 3—DASS 2 and DPNSS only (A/B end, X/Y end)
- Rate adaption protocol—DASS 2 and DPNSS only (X.30)

**Note:** The TAOS unit can receive multichannel calls using MP encapsulation only if all channels of the call share a common telephone number (namely, a hunt group). You can request that your service provider supply you with a hunt group.

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