SL-1

Electronic Switched Network

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Introduction

The Electronic Switched Network (ESN) is a private communications network intended for use by large business customers with distributed operating locations.

This document introduces the reader to the concepts of the ESN with emphasis on the switching components.

References

For more information on the Electronic Switched Network (ESN) or on networking features and load configuration, refer to the following Northern Telecom publications:

- *Electronic Switched Network signaling guidelines* (309-3001-180)
- *Electronic Switched Network transmission guidelines* (309-3001-181)
- *X11 features and services* (553-3001-305)
- *Basic and Network Alternate Route Selection description* (553-2751-100)
- *Coordinated Dialing Plan description* (553-2751-102)
- *ISDN Primary Rate Interface description and administration* (553-2901-100)
- *ISDN Primary Rate Interface installation* (553-2901-200)
- *ISDN Primary Rate Interface maintenance* (553-2901-500)
- *X11 input/output guide* (553-3001-400)

The prime element of the Electronic Switched Network (ESN) is an ESN node. The ESN nodes are strategically located (see Figure 1) to concentrate on on-network traffic and access to off-network facilities efficiently and economically. An ESN can be a single ESN node serving a few locations in a metropolitan area or multiple ESN nodes serving up to 999 locations that can be located across the country.

The ESN nodes direct calls from a switch in one geographical location to a switch in any other geographical location in a cost-efficient and easy-to-use manner by:

- eliminating long, complex dialing plans and replacing them with an abbreviated Uniform Dialing Plan (UDP) common to all switches that are part of the ESN
- providing a means of controlling the number and types of trunks that are available to each network user and the Time of Day that access to a trunk (or group of trunks) is allowed
- selecting the least-cost trunk route available to complete a call between switches automatically
- providing uniform network access to stations served directly by the ESN node and stations served at other switches (ESN mains, Conventional mains) connected by means of tie trunks to an ESN node as shown in Figure 1.
- providing the call originator with the option to either accept or refuse a call that is to be completed over an expensive trunk route if there are not any least-cost trunk routes currently available
- providing optional queuing features that allow the call originator (when all trunks are currently busy) either to remain off hook until a trunk becomes idle or to hang up and receive a call back from the SL-1 when a trunk becomes idle — thus eliminating repeated, time-consuming dialing attempts
- collecting and printing traffic data related to the ESN operation (thus allowing the efficiency of the network to be monitored)

To accomplish the efficient call-handling operations in an ESN, each ESN node utilizes some, or all, of the following unique ESN software features listed below and described in this document:

- Transparent Data Networking (TDN, described in *Transparent Data Networking* (553-2731-110), which provides a transparent data channel for data modules to perform end-to-end protocol exchange. With TDN, two data modules wait for a circuit path to be established before exchanging protocol parameters.
- Network Class of Service (NCOS)
- Network Alternate Route Selection (NARS)
- Network Signaling (NSIG)
- Network Traffic Measurements (NTRF)
- Off-Hook Queuing (OHQ)
- Call-Back Queuing (CBQ)
- Coordinated Call-Back Queuing (CCBQ)
- Call-Back Queuing to Conventional Mains (CBQCM)
- Free Calling Area Screening (FCAS)
- Coordinated Dialing Plan (CDP)
- Network Authorization Codes (NAUT)

Switch definitions

In the context of this publication, the following definitions of the various switch types that are considered to be part of an ESN network can be applied. These definitions are used for convenience and do not constitute the only possible ESN configuration.

ESN node A switch equipped with features identified in Table 1 and configured within the network as shown in Figure 1.

ESN main A switch equipped with features identified in Table 1 and that is connected by means of tie trunks to a single ESN node. An ESN main can also be equipped with the Basic Alternate Route Selection (BARS) feature to provide alternate route selection capabilities for calls placed to satellite switches that are located on the ESN main. See *Basic and Network Alternate Route Selection description* (553-2751-100).

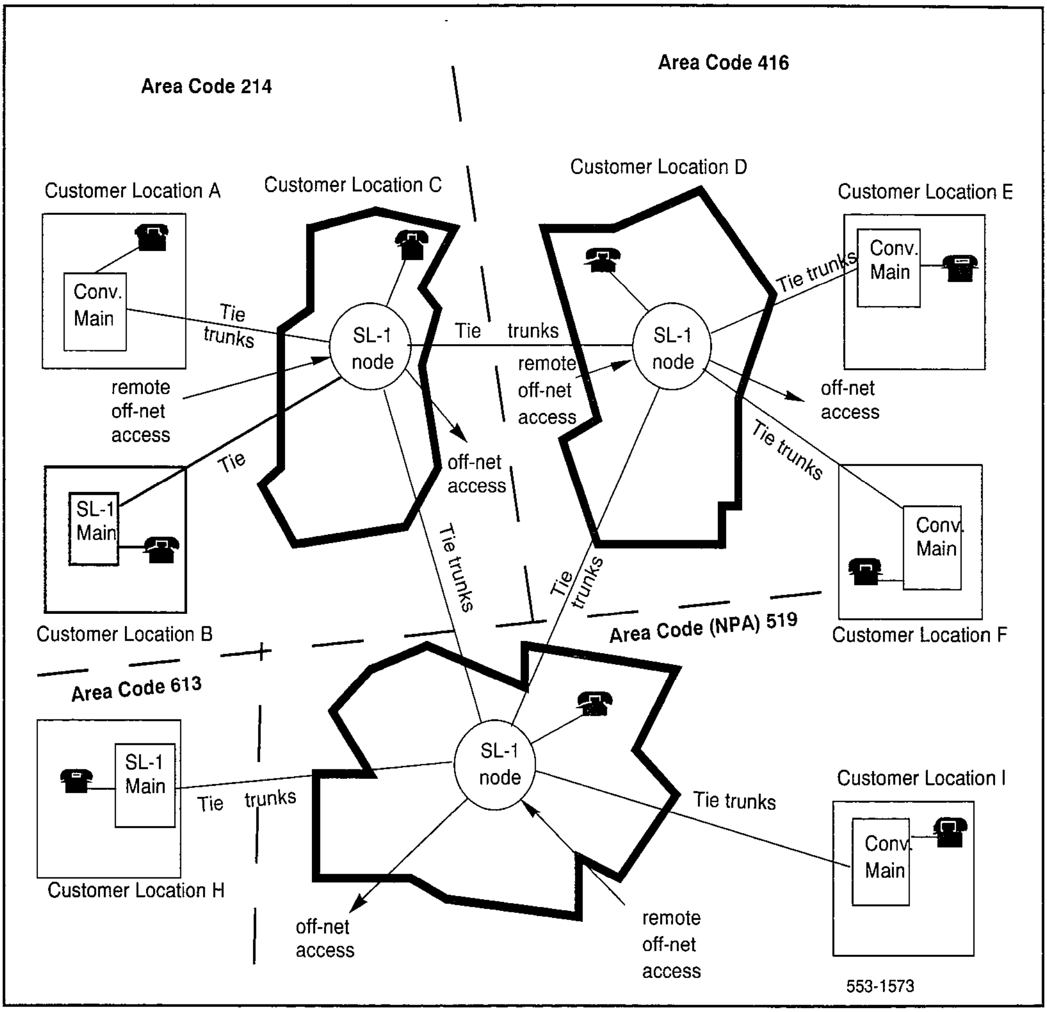
Conventional main A switch that is connected to an ESN node and equipped with none of the features listed in Table 1. Other switch types can include Step-by-Step (SXS), Electronic Tie Network (ETN), and DIMENSION.

Table 1
Feature package requirements for ESN nodes and ESN mains

Description	ESN node	ESN main
Network Alternate Route Selection	X	
Off-Hook Queuing	X	
Call-Back Queuing	X	
Coordinated Call-Back Queuing	X	X (optional)
Call-Back Queuing to Conventional Mains	X	
Network Signaling	X	X
Free Calling Area Screening	X	
Network Class of Service	X	X
Coordinated Dialing Plan	X	X (optional)
Basic Alternate Route Selection		X (optional)
Network Authorization Code	X	
Basic Authorization Code		X (optional)
Network Traffic Measurements	X	X (optional)

Note: A switch is termed a Conventional main if none of the above feature packages are equipped at the switch.

Figure 1
Example of an Electronic Switched Network



Network Class of Service

The Network Class of Service feature (NCOS) is an integral part of Network Control (NCTL) and Routing Control in an ESN. NCOS provides the means to control:

- which trunk routes are eligible for access to attempt call completion
- whether or not queuing is offered to a call originator
- whether or not the originator of a network call receives a warning tone when an expensive trunk is selected to complete the call
- whether or not the user is allowed to access the Network Speed Call (NSC) feature

An ESN node (and ESN main) can accommodate 100 (0 to 99) to Network Class of Service (NCOS) groups, with each group made up of different network-access characteristics. Once each NCOS group is defined through service change, the line, trunk, and attendant groups connected directly to the switch are assigned to the NCOS group that best serves the particular requirements of each. The NCOS to which each group is assigned is independent of the regular class of service assigned to them. Incoming tie trunks that connect another switch (ESN main, Conventional main, ETN switch) to the ESN node are also assigned to an NCOS group that determines their level of access to the network facilities at the ESN node.

> *Note:* Pre X11 release 13, an ESN node can accommodate 16 (0 to 15) NCOS groups.

Facility Restriction Level

Included as part of each NCOS group is a Facility Restriction Level (FRL) number that ranges from 0 (low-privilege) to 7 (high-privilege). The FRL is used by the software to determine the alternate route selection choices available for specific network call attempts by a line or trunk within an NCOS group.

For example, a user assigned in an NCOS group having an FRL of 3 would be allowed access only to alternate route selection choices assigned an FRL of 3 or less; access to trunks with an FRL greater than 3 would be denied. Thus, by assigning low-privilege network users to an NCOS group having a low FRL and high-privilege network users to an NCOS group having a higher FRL, the customer can control access to all network facilities.

Expensive Route Warning Tone

In some instances, expensive trunk routes can be assigned to an NCOS group with an FRL that would allow them to be accessed by some network users. When this occurs, the originator of the network call may be sent an optional Expensive Route Warning Tone (ERWT).

The ERWT tone alerts the caller that an expensive route has been selected to complete the call and provides the caller with the option of either accepting or rejecting the call before it is completed over the expensive route. Eligibility for ERWT is allowed or denied to individual lines and incoming trunk groups on an NCOS group basis.

Queuing

Each NCOS group also defines whether or not the various queuing features are available to lines or trunks assigned to the group.

Network Alternate Route Selection

Network Alternate Route Selection (NARS) provides a comprehensive and flexible networking package that can be tailored to a customer's network. Prime elements of the NARS feature are:

- simple network access codes
- a Uniform Dialing Plan (UDP)
- dialing transparency
- automatic least-cost routing
- Time of Day (TOD) routing
- automatic on-network (on-net) to off-network (off-net) overflow
- network controls through Network Class of Service (NCOS) and Facility Restriction Levels (FRL)
- One to ten digit translation, restriction, recognition (X11 release 5 and later)
- Free Calling Area Screening (FCAS)
- Expensive Route Warning Tone (ERWT)

Table 2 lists the features equipped for Network Alternate Route Selection/Basic Alternate Route Selection (NARS/BARS). Parenthetical values are for releases prior to X11 release 13. If NARS/BARS features are equipped on the same switch for different customers, the NARS values apply to the switch.

Table 2
NARS/BARS feature parameters

Parameter	Features equipped at switch	
	BARS	**NARS**
NCOS groups	0–99 (0–7)	0–99 (0–15)
Facility Restriction Levels	0–7	0–7
Digit manipulation tables	1–255	1–255
Route lists	0–127	0–255
Route list entries	0–31 (0–7)	0–31 (0–7)
FCAS tables	1–127	1–255
SDR tables	0–255	0–511

Legend:
NCOS = Network Class of Service
FCAS = Free Calling Area Screening
SDR = Supplemental Digit Restriction

If the New Flexible Code Restriction (NFCR) feature (see *X11 features and services* (553-3001-305)) operates in conjunction with Basic Alternate Route Selection (BARS) and/or Coordinated Dialing Plan (CDP), the number of available NCOS groups is eight for releases prior to X11 release 13 and 100 beginning with X11 release 13.

NARS access codes

To access Network Alternate Route Selection (NARS), the user at an ESN node, ESN main, or Conventional main dials either one of two customer-assigned network access codes, AC1 or AC2. These access codes are typically "8" for on-net and long distance calls (AC1), and "9" for off-net and local calls (AC2). However, any one- or two-digit code can be used, provided the access code assigned for AC1 is different from that assigned for AC2 and there is no conflict with any other part of the dialing plan.

> *Note:* Only TIE trunks allow digit insertion on the AC1 code. Direct Inward System Access (DISA) Central Office (CO) trunks require the user to dial the AC1 code.

Dialing a NARS access code triggers NARS to perform the necessary call processing and routing using a specified set of network translation tables. This mechanism is used to implement the Uniform Dialing Plane (UDP) for private networks. (NARS dial tone may or may not be provided to the caller after an access code is dialed, at the option of the customer.)

Uniform Dialing Plan

The Uniform Dialing Plan (UDP) enables users at an ESN node, ESN main, or Conventional main to dial all calls in a uniform manner, regardless of the location of the calling party or the route that the call takes.

UDP for on-net calling

An on-net call is one that terminates at a customer-owned location. To reach any on-net location, the user dials the on-net access code (AC1), followed by seven digits. The format for this call would be:

AC1 * LOC + XXXX

Legend:

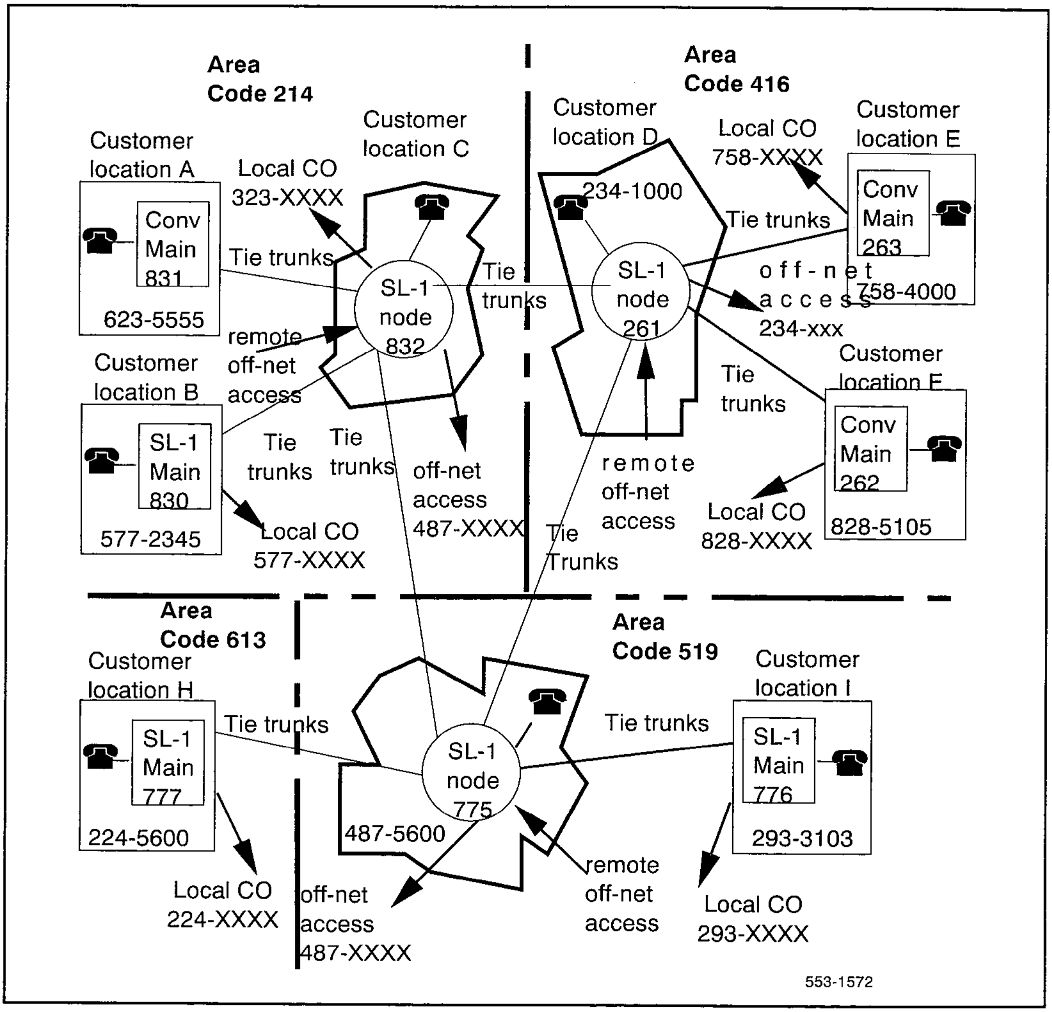
AC1	=	the one-digit or two-digit on-net access code
*	=	pause for NARS dial tone (optional)
LOC	=	a three-digit Location Code assigned for the destination location (area code)
XXXX	=	the extension number of the party to be reached at the destination location

Each switch that is part of the network (including the ESN nodes) is identified by a unique three-digit Location Code (LOC) assigned at the ESN node. If 1+ dialing is used, there must be no conflict between the Location Code number assigned for a switch and all Numbering Plan Area (NPA) codes. (Optimally, 1+ dialing applies to NPA calls but not to local or network calls).

A customer-owned location can be either physically connected to the network (by means of private trunk facilities) or virtually connected to the network (by means of public facilities). If a location is virtually connected to the network, the dialed Location Code (LOC) is translated and converted by NARS translation (at the ESN node) into the public number for the virtual location; for example, the Direct Distance Dialing (DDD) number or the Direct Inward Dialing (DID) number.

Figure 2 illustrates an ESN network with a typical UDP. A user at any customer location, such as LOC 776 (Conventional main, location I) who wishes to call extension number 3283 at LOC 777 (ESN main, location H) must first dial 8 (AC1), pause for the optional NARS dial tone from ESN node at location G, then dial 777-3283.

Figure 2
Example of an ESN with a typical Uniform Dialing Plan



UDP for off-net calling

An off-net call is one that does not terminate at a customer-owned location, although network facilities may handle some of the call routing. As Figure 2 indicates, a call is off-net if a user at LOC 776 calls a number associated with Central Office (CO) 758-XXXX in the foreign area code 416.

Table 3 lists the dialing formats for the various types of UDP calls.

UDP for switch equipped with Directory Number Expansion

The Directory Number Expansion (DNXP) package allows an internal Directory Number (DN) to have up to seven digits. If the DNXP package is equipped, a seven-digit LOC call to an ESN switch can be terminated to a seven-digit internal DN. This is accomplished by using the Digit Manipulation Index (DMI) for a Home Location Code (HLOC). The HLOC can have an optional DMI that defines digits to be inserted and leading digits to be deleted. The DMI option can be selected and defined in LD90 (refer to *X11 input/output guide* (553-3001-400)).

A seven-digit LOC call to an ESN switch with five-digit, six-digit, or seven-digit internal Directory Numbers (DNs) can be achieved by manipulating its Home Location Codes (HLOCs) – deleting the first three digits and inserting one, two, or three digits respectively.

For example, an existing ESN switch with LOC 646 and four-digit dialing plan is upgraded to a five-digit dialing plan by prefixing the digit 2. An incoming on-net (LOC) call can be handled as follows:

1 Define a Digit Manipulation Index (DMI) for the Home Location Code (HLOC) 646 by:
   - Deleting the first three digits (for example, 646)
   - Inserting the digit 2

2 The Location Code (LOC) call 646-2359 to a telephone of this switch has become a HLOC call in the switch (with the same digit string). Therefore, it is converted to 22359 and terminated to the desired telephone.

Table 3
Dialing formats for NARS UDP calls (Part 1 of 2)

Call type	Dialing format	Code capacity
On-Net (Notes 1, 2 and 3)	AC1 * LOC + XXXX	640
DDD FNPA (Note 1)	AC1 * 1 + NPA + NXX + XXXX	160
Network Speed Call	AC1 * LA + LN	1
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX	160
International DDD	AC1 * 011 + CC + NN	99
Operator-assisted International DDD	AC1 * 01 + CC + NN	99
DDD HNPA (Note 1)	AC2 * 1 + NXX + XXXX	1
DDD operator	AC2 * 0	1
Local calls (Note 1)	AC2 * NXX + XXXX	640
Special local services	AC2 * SPN	8
Toll-free calls	AC2 * 800 + NXX + XXXX	1
Toll-free calls (Note 1)	AC2 * 1 + 800 + NXX + XXXX	1
Toll special numbers	AC2 * 900 + NXX + XXXX	1
Toll special numbers (Note 1)	AC2 * 1 + 900 + NXX + XXXX	1

AC1 Access code for on-net, long distance and Network Speed Calls. Can be any 1 or 2 digits number

AC2 Access code for local calls. Typically the digit "9," can be either 1 or 2 digits in length.

Note 1: If 1+ Dialing is used, the On-Net and Local Calls code capacities are increased to 800 and 792 respectively.

Note 2: If the code 1XX is reserved for future 1+ Dialing use, and not for Network Speed Call codes, then the location code capacity will be reduced to 639 if a 3-digit NSC code is used, 632 if a 2-digit NSC code is used, or 560 if a one-digit NSC code is used.

Note 3: When 1+ Dialing is used, Network Speed Call access will be in the form of 2XX–9XX as a subset of the location codes utilized in the UDP. The location code capacity will be reduced to 799 if a 3-digit NSC code is used, 792 if a 2-digit NSC code is used, or 720 if a one-digit NSC code is used.

Table 3
Dialing formats for NARS UDP calls (Part 2 of 2)

Call type	Dialing format	Code capacity
*	Symbol meaning wait for NARS dial tone (optional).	
NPA	Numbering Plan Area (NPA) code. Any number of the form NXX.	
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form NXX.	
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form NXX.	
CC	Country code. Any 1, 2, or 3 digits from 2 to 9.	
NN	National Number. Depends on national dialing plan; max 12 digits including Country Code.	
N	Any of the digits 2 to 9.	
X	Any of the digits 0 to 9.	
LA	List access code. Any 1, 2, or 3 digits from 0 to 9.	
LN	List element number. Any 1, 2, or 3 digits, maximum of 1000 element numbers.	
LOC	Three-digit location code for each UDP network location.	
NXX	Local Exchange Code.	
XXXX	4-digit directory number.	
SPN	Special numbers: for example, 411, 611, or may be XXXX.	

Note 1: If 1+ Dialing is used, the On-Net and Local Calls code capacities are increased to 800 and 792 respectively.

Note 2: If the code 1XX is reserved for future 1+ Dialing use, and not for Network Speed Call codes, then the location code capacity will be reduced to 639 if a 3-digit NSC code is used, 632 if a 2-digit NSC code is used, or 560 if a one-digit NSC code is used.

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Dialing transparency

Extending the UDP to a remote-access switch (ESN main or Conventional main) is accomplished by forming a single tie trunk access group between the remote switch and the ESN node. Users at this remote switch access the trunk group to the node by dialing the on-net access code (AC1). The ESN node is arranged to insert the digit for AC1 on each incoming call from the switch automatically, this allows access for on-net and long distance calling in a transparent fashion. Local calling is arranged through conventional dial "9" Central Office (CO) trunks at the remote access switch.

> *Note:* If an ESN node replaces a tandem switch in a Tandem Tie Trunk Network (TTTN), other tandem switches in the network can "tandem through" the ESN node using the same access codes as before. This requires that there be no dialing conflicts between the access codes for the TTTN trunks and the dialing plan implemented at the ESN node.

Automatic least-cost routing

For each call translated at an ESN node, NARS selects a route entry from a route list of up to 32 outgoing alternate routes (eight alternates prior to X11 release 13) to complete the call. Any combination of trunks, for example, Central Office (CO), Foreign Exchange (FX), or TIE, can be specified in a route list.

> *Note:* The Meridian Bandwidth Controller (DCA System 9000), in conjunction with ESN routing lists, improves the capacity of T1 based tandem networks by choosing the optimum path for a voice or data call.

Typically, the first entries in a route list are the least-cost routes to a destination and are defined through a service change as the initial set (I set). The remaining routes in the list (if any) are the more expensive routes to a destination and they are the extended set of routes in the list. Refer to "Call-Back Queuing" on page 45 and "Off-Hook Queuing" on page 51 for more information on I set and extended set routes.

Associated with each entry in a route list is information relevant to:

- the route number (0 to 511)
- the minimum Facility Restriction Level (FRL) required for access
- the time of day the route can be accessed
- whether or not queuing Call-Back Queuing (CBQ) or Off-Hook Queuing (OHQ) is allowed on the route
- whether or not the route is to receive Expensive Route Warning Tone (ERWT) treatment
- a digit manipulation table index number (0 to 255)
- a Free Calling Area Screening (FCAS) table index number (0 to 255)
- information for conversion from an on-net call to an off-net call

Route eligibility

NARS translates a dialed LOC, Numbering Plan Area Code (NPA), Local Exchange Code (NXX), or Special Number (SPN) into a route list and searches the list sequentially for an available route. Route eligibility for a given call is based on the caller's NCOS, the NCOS-defined Facility Restriction Level (FRL), the current Time of Day (TOD), and Class of Service (CLS).

Because each entry in a route list has a minimum FRL required for access and all network users are assigned an FRL through their NCOS, the network communications manager can restrict the type of calls allowed to users.

For example, a user assigned to an NCOS group with an FRL of 0 would only be able to make calls to the special local services numbers that have an FRL of 0. In addition, the communications manager can restrict high-cost facilities by assigning a high FRL to the expensive routes in a route list and a lower FRL to a user's NCOS.

Digit manipulation

Any trunk type can be specified in a route list. However, certain trunk types require that Network Alternate Route Selection (NARS) modify dialed digits to conform to trunk dialing requirements. To do this, NARS uses digit manipulation tables to modify the dialed digits.

Each digit manipulation table, up to 256, is associated with a Digit Manipulation Index (DMI) number and defined at each ESN node as shown in Figure 3. Digit manipulation can delete up to 15 leading digits and insert up to 20 leading digits.

See Figure 2 for an example of an ESN with a typical Uniform Dialing Plan. A user at Conventional main location I dials the number 8-613-596-9084 to reach an off-net station in the 613 Numbering Plan Area (NPA) associated with ESN main, location H.

At the ESN node, NARS selects the appropriate route list for call completion to NPA 613 and finds that the only available route to that NPA is a local CO trunk that requires the insertion of the leading digit "1" for long distance calls. The route list entry for this route specifies a Digit Manipulation Index (DMI) number (0 to 255; "0" means no digit manipulation is required). NARS references the digit manipulation table indicated by the index number, deletes digits as specified in the table (none in this case), inserts the required digits ("1" in this case) and completes the call on this route.

Time of Day routing

NARS provides up to eight (0 to 7) Time of Day (TOD) schedules. Each entry in a route list is assigned to the TOD schedule that specifies the hour(s) that the particular entry can be accessed to ensure the most cost-effective route selection. A typical TOD schedule is shown in Table 4. A TOD schedule can be associated with any number of arbitrarily selected 15-minute periods. However, any one 15-minute period can appear only in one TOD schedule.

Based on the Time of Day (TOD) schedule in Table 4, a route list entry assigned to TOD schedule 2 is accessed only between the hours of 00:00 to 07:44 and 17:30 to 23:59. Access to the route at any other time is denied. TOD schedules can be turned "on" or "off" through service change as the traffic conditions change. A TOD schedule is turned on for an entry by turning off all other TOD schedules.

Table 4
A typical TOD schedule

TOD schedule	Time period
2	00:00 to 07:44 17:30 to 23:59
1	07:45 to 08:59 12:00 to 13:14 16:00 to 17:29
0	09:00 to 11:59 13:15 to 15:59

Flexible ESN "0" Routing

Flexible ESN "0" Routing, available beginning with X11 release 16, uses four prefixes to call the local operator (prefix of 0) or international operator (prefix 00), or to make station-to-station international calls (011), calling card, collect, or other operator-assisted international calls (01).

Normally, the ESN translation table only contains leftwise unique numbers (for example, if one entry begins with the digits "123," no other entry can begin with the digits "123." The four special "0" prefixes, which are not leftwise unique, are an exception to this rule.

Flexible ESN "0" Routing is part of the existing Basic Alternate Route (BARS) (57) and NARS (58) packages and interacts only with these features. Since NARS has two translation tables, two Flexible ESN "0" Routing data blocks are included in NARS. A call could be routed two different ways.

Flexible ESN "0" Routing applies to all route types and network types that are supported by ESN. For information on the appropriate prompts and responses in service change (LD90), refer to Northern Telecom Publication *X11 input/output guide* (553-3001-400).

Automatic on-net to off-net overflow

If all on-net facilities to a location are busy or blocked, NARS can convert a dialed UDP number to the Listed Directory Number (LDN) or Direct Inward Dialing (DID) number of the destination location and use off-net facilities to complete the call.

For example, a user at Conventional main location I (Figure 2) dials 8-777-3283 to reach a party with extension number 3283 at ESN main location H. At the ESN node, NARS translates the dialed LOC number (777) into a route list, and searches all eligible routes in the list. Failing to find an available tie trunk route, NARS then seizes local off-net facilities and, to complete the call, outpulses either:

- 224-3283, if location H is arranged for Direct Inward Dialing (DID), or
- 224-5600, if location H is not arranged for Direct Inward Dialing (DID)

The overflow feature is subject to these restrictions:

- Only one Listed Directory Number (LDN) may be defined per Location Code (LOC).
- DID numbers must have the same Local Exchange (NXX) as the LDN.
- Only one contiguous DID Directory Number (DN) range can be defined per location. DNs that lie outside the range are converted to the LDN.

Multiple DID Office Code Screening

Multiple DID Office Code Screening is available with X11 release 5 and later as a NARS overflow enhancement. The screening process helps route network calls through the public network using on-net to off-net conversion. The call can terminate at any DN defined in the Location Code memory data block.

For each LOC defined, Multiple DID Office Code Screening allows:

- the definition of multiple Local Exchange (NXX) codes
- the definition of multiple ranges of DN within each NXX

The following arrangements of multiple office codes (NXX) and multiple DN ranges are possible:

- single office code with a single DN range (the only alternative prior to X11 release 5)
- single office code with multiple DN ranges
- multiple office codes, each with a single DN range
- multiple office codes with multiple DN ranges

Multiple DID Office Code Screening is subject to the following requirements:

- Only one NPA per LOC is allowed.
- Ranges defined within an LOC must be unique. Overlapping or duplication of ranges is not permitted.
- Each DID range is limited to four digits.
- A maximum of 20 DID ranges may be defined per LOC, regardless of the number of office codes.

Incoming Trunk Group Exclusion

With X11 release 5 and later, Incoming Trunk Group Exclusion (ITGE) enhances the NARS feature that blocks calls from main users trying to reach destinations in the Home Numbering Plan Area (HNPA) or other restricted NPAs, NXXs, LOCs and Special Numbers (SPNs). When the feature is configured, users cannot use the network to circumvent the restrictions. They are forced to dial off-net instead and become subject to whatever restrictions are imposed at the main.

Standard call blocking affects outgoing calls to a specific NPA, NXX, Special Number (SPN) or LOC at the ESN node if the call is from a specific incoming trunk group. As a result:

- The caller cannot loopback through a home switch, for example, the Home Numbering Plan Area (HNPA) or home NXX. Calls that should have been made off-net from the caller's home switch are blocked at the node.
- Main users cannot use ESN to make calls to NPA, NXX, Special Number (SPN), or LOC numbers that they are not permitted to call from the home switch.

Customers define their own sets of restricted trunk routes to specific NPA, NXX, SPN or LOC. There is one Incoming Trunk Group Exclusion Index (ITEI) (maximum 255) for each defined NPA, NXX, SPN or LOC. Each index points to an Incoming Trunk Group Exclusion (ITGE) table that defines a maximum of 128 restricted routes. Incoming Trunk Group Exclusion provides full ten-digit restriction for NPA and SPN codes, seven-digit restriction for NXX codes and LOC. The code itself may be restricted.

NARS tests every ESN call it receives to see if the dialed code is a restricted SDRR type (Supplemental Digit Restriction/Recognition). If it is, NARS checks whether or not it has an ITGE restriction and if there is an Incoming Trunk Exclusion Index (ITEI) number (1 to 255) associated with it. If an ITEI is defined, NARS searches the ITGE table corresponding to the dialed code. If the incoming trunk route is a member of the ITGE, the NARS process terminates and the call is blocked. Otherwise, call processing continues.

Off-Net Number Recognition

Off-Net Number Recognition eliminates the need to use two extra CO trunks when a private network subscriber dials a DID or Direct Distance Dialing (DDD) number that terminates at an ESN location. Calls are routed directly to the dialed DN (DID calls) or to the LDN (DDD calls), rather than being switched from the terminating ESN switch to the CO and back again.

The customer defines Off-Net Number Recognition parameters for local and remote DDD and DID locations in the Network Translation Tables and Supplemental Digit Restriction/Recognition blocks (SDRR). Recognition of up to ten digits can be defined.

Table 5
Network Translation Tables

Call type	Network translation table (number of digits)	SDRR block (number of digits)
NPA	3	1–7
NXX	3	1–4
SPN	4	1–7

Up to 512 Supplemental Digit Restriction/Recognition (SDRR) blocks can be defined. Each table can contain up to 64 entries. Off-net numbers are recognized at the last intelligent NARS/BARS switch. Translation of the NPA, NXX or SPN identifies the method of treatment for the call. If the data type is SDRR and the index is an SDRR table index, Supplemental Digit Restriction/Recognition is applied by comparing the dialed digits with the numbers declared in the SDRR block.

- If no match is found in the SDRR, route selection is called, call processing resumes and the call is routed to the CO of the terminating off-net number.
- If a match is found and the number is in the "denied" block, standard call blocking occurs.
- If a match is found and the number is recognized as a terminating number at the local switch (for example, the last intelligent NARS/BARS switch), the call is terminated at DN (DID calls) or at the attendant DN (DDD calls).

– If a match is found and the dialed number is a recognized number terminating at a remote Conventional main, route selection is called, the appropriate digit manipulation takes place, and the call is routed directly to the conventional main. DID calls terminate at the dialed number and Direct Distance Dialing (DDD) calls terminate at the attendant DN.

Digit translation/restriction/recognition

The ESN provides a 1-digit through 11-digit translation/restriction/ recognition capability through the use of Network Translation Tables. There are two Network Translation Tables, one associated with each of the network access codes (AC1 and AC2).

Normal translation mechanisms translate the dialed network access code, determine that the call is to be processed by NARS, and select the appropriate Network Translation Table as shown in Table 5. The NARS translation determines the method to be used to process the call and applies digit restriction or recognition where required. The result of translation is to invoke route selection via a specified route list or to block the call. More information on digit recognition can be found under "Off-Net Number Recognition" on page 24.

Network Translation Table

Each NPA (area code) translation entry, excluding the Home Numbering Plan Area (HNPA) contains:

– a route list index number (0 to 255) that indicates which route list to use for a call to this NPA

– whether or not there are telephone numbers within this NPA to which network calls are to be blocked

– whether or not there are telephone numbers within this NPA that cannot receive network calls because of ITGE restrictions (X11 release 5 and later)

– whether or not there are numbers under this NPA that are to be recognized as DID or Direct Distance Dialing (DDD) codes to an on-net location (X11 release 5 and later)

– a list (up to 64) of one- to seven-digit numbers that follow the NPA and are to be blocked or recognized in this NPA

Each NXX office code translation entry contains:

- a route list index number (0 to 255) that indicates which route list to access in processing a call to this NXX
- whether or not there are telephone numbers within this NXX to which network calls are to be blocked
- whether or not there are telephone numbers within this NXX to which network calls are to be blocked because of ITGE restrictions
- whether or not there are numbers under this NXX that are to be recognized as DID or DDD codes to an on-net location
- a list of up to 64, one- to four-digit numbers that follow the NXX and are to be blocked or recognized in this NXX

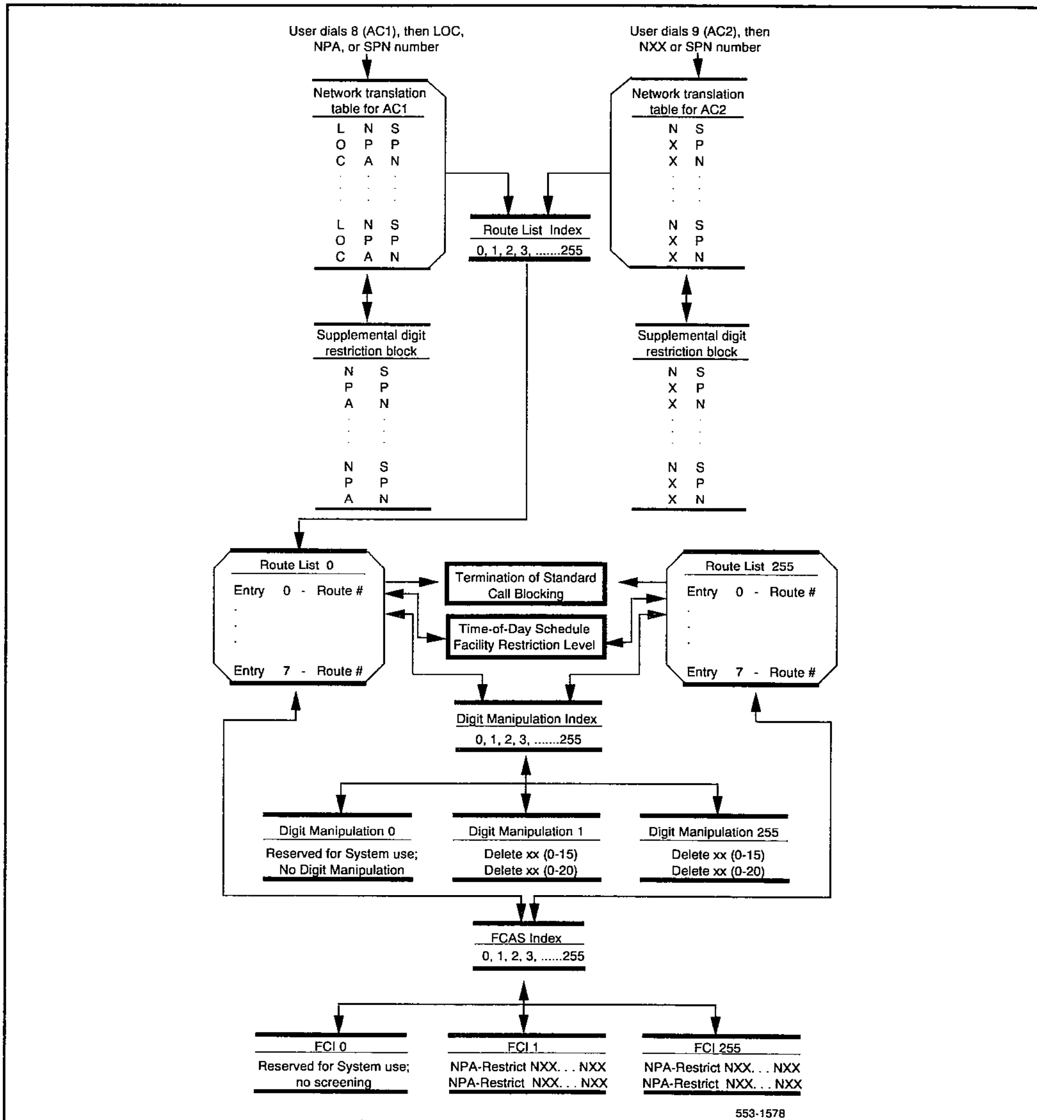
Each LOC translation entry, excluding the HLOC contains:

- a route list index number (0 to 255) that indicates which route list to access in processing a call to this LOC
- the Listed Directory Number (LDN) to which the LOC number is to be converted when using off-net DDD facilities
- the ranges of DID numbers to which the LOC number can be converted when using DID facilities
- whether or not there are LOC numbers to which network calls are to be blocked because of ITGE restrictions

Each Special Number (SPN) translation entry contains:

- a route list index number (0 to 255) that indicates which route list to access in processing a call to this SPN
- whether or not there are digits following SPN numbers to which network calls are to be blocked
- whether or not there are SPN numbers to which network calls are to be blocked because of ITGE restrictions
- whether or not there are SPN numbers that are to be recognized as codes to an on-net location
- a list of up to 64 entries, of one- to seven-digit numbers that are to be blocked or recognized when following the SPN

Figure 3
NARS elements accessed at an ESN node to process a network call



Supplemental digit restriction

Supplemental digit restriction blocks, which vary by release, function as follows:

- block access to certain telephone numbers
- recognize off-net calls dialed to on-net locations
- prevent routing of calls to the home switch of the originating trunk group by either on-net or off-net facilities

See Table 6 for more information. The customer can also specify through service change the treatment that blocked calls receive. For example, overflow tone, intercept to attendant, or recorded announcement. One block can be assigned per NPA, NXX, or SPN.

Table 6
Supplemental digit restriction blocks per NARS or BARS

Network package	X11 release 4	X11 release 5 and later
ESN node	256	512
BARS	32	256

Note: For X11 release 4 and earlier, one block can recognize or restrict up to 16 numbers. For X11 release 5 and later software, one block can recognize or restrict up to 64 numbers.

Free Calling Area Screening

The Free Calling Area Screening (FCAS) is a NARS feature that provides the customer with the capability of full six-digit (NPA-NXX) screening to determine the route choice for completion of off-net calls. With FCAS, a customer can allow calls to NXX codes within the "free calling area" surrounding a particular on-net location, and restrict (deny) calls to those NXX codes that would incur long distance charges.

The FCAS is implemented in a similar fashion to digit manipulation, for example, through FCAS tables. There can be up to 256 FCAS tables defined at an ESN node. Each table can contain up to 15 NPA codes. Up to 800 NXX codes can be restricted within each NPA code. Each FCAS table is referenced by a Free Calling Index (FCI) number (0 to 255), where "FCI = 0" is a system default meaning no Free Calling Area Screening is required. The appropriate FCI number is then assigned to the applicable route list entries.

Whenever a route list entry is being considered for an off-net call (for example, 8-NPA-NXX-XXXX), NARS checks to see if there is an FCI number (other than 0) referred to by the entry. If an FCI number other than 0 is defined, the appropriate FCAS table for the dialed NPA is found and used for NXX screening.

If the dialed NXX is denied in the table, NARS does not use the route list entry for call completion, but continues to search for another eligible route list entry. If the dialed NXX is not denied in the table, the route list entry is eligible for the call. Calls converted to the LDN of a location are screened only if the NPA is included as part of the LDN. NXXs allowed in an Free Calling Index (FCI) table are the only ones allowed for that route list entry.

Expensive Route Warning Tone

This feature enables the network manager to select certain users to receive an Expensive Route Warning Tone (ERWT). Eligibility for this tone is based on the user's NCOS. The tone, which is three 256-ms bursts of 440 Hz, notifies the user that NARS has selected facilities designated as expensive to complete the call. Upon receipt of ERWT, the user has the choice of either allowing the call to complete over the expensive facilities or going on hook to avoid the increased expense. The user must make this choice within a programmable time of 0 to 10 seconds. The tones must be activated for the customer group and the expensive route cannot be part of the initial set (I set) in the route list.

If the call originator is located at an ESN node or ESN main, the Ring Again (RGA) feature is defined for the user, and the user is eligible for extended CBQ option ([a]), then Ring Again may be activated to queue the call; see the various queuing features for more information.

If the ESN node is equipped for Call Detail Recording (CDR), acceptance of an expensive route after ERWT is received is noted in the CDR record.

NARS bypass control

A customer can allow selected users to bypass the NARS feature for call completion between any two locations, for example, two locations that share a high community of interest. To do this, routes and trunks are set up between the two locations and assigned an access code distinct from the AC1 and AC2 codes used to access NARS.

The normal trunk controls; for example, Trunk Group Access Restriction (TGAR), Class of Service (CLS), and code restriction are then used to enable access only to the selected users. All other users are denied access to the trunk group and are forced to use NARS for all calls.

Network speed call

The Network Speed Call (NSC) feature enables a user at an ESN node, ESN main, or Conventional main who is normally restricted from making certain types of NARS calls to make such a call if the destination is a company-approved number defined in a System Speed Call (SSC) list. This feature requires that the System Speed Call feature be equipped in addition to Network Speed Call. See *X11 features and services* (553-3001-305).

The user dials the NARS access code, then dials a customer-defined Network Speed Call (NSC) access code (one, two, or three digits) to make a speed call. The NSC access code must be different from Special Numbers and from all LOC, NPA, and NXX codes.

A service change in the network translation overlay associates an NSC access code with a System Speed Call (SSC) list (maximum number of lists is 4096 beginning with X11 release 13; earlier, the limit was 254). If the SSC list changes in length, the list access code and list number must be deleted and reentered into the NARS translator. An NCOS number is associated with each SSC list, but applies to the call only if the FRL (0 to 7) is greater than that associated with the call originator's assigned NCOS.

> *Note:* If 1+ dialing is specified for an NPA, NXX, or SPN number in a translator, the digit "1" must not be used as the leading digit for Network Speed Call list codes in that translator.

The user then dials the number of the desired entry in the SSC list. Entry numbers can range from 0 to 9, 00 to 99, or 000 to 999, depending upon the number of elements allocated when the list was defined through service change. After the dialing is finished, the digits defined for the list entry are passed to NARS/BARS translation for processing. Route and feature (Off-Hook Queuing [OHQ], Call-Back Queuing [CBQ]) eligibility for call completion are based on the FRL of the NCOS associated with the SSC list only if the FRL of the user's assigned NCOS is lower than that of the list.

Network Call Transfer

Network Call Transfer (NXFER) improves the operation of the existing Call Transfer (XFER) feature between two switches when a call is transferred back to the originating switch. The regular XFER feature requires two tie trunks to complete the call. With NXFER, the originating switch completes the transfer itself, and the tie trunks are dropped. Refer to *ISDN Primary Rate Interface description and administration* (553-2901-100) for a detailed description.

The benefits of the Network Call Transfer (NXFER) feature include:

- reduced use of access tie lines
- improved transmission performance, since tie lines are not used for the completed connection
- similar operation to the existing XFER feature

Figures 4 and 5 illustrate a Network Call Transfer (NXFER) operation in which telephone A receives a call from B and transfers it to C. As shown in Figure 4, NXFER and Network Signaling (NSIG) software at both ends of the tie trunk allow telephone A at one ESN switch (I) to transfer the tie trunk call from telephone B (switch II) to a third party, telephone C (switch II). As shown in Figure 5, if the transfer is allowed, stations B and C connect on switch II. The ESN tie trunks are dropped after the transfer occurs. In comparison, regular Call Transfer (XFER) requires two tie trunks and both switches to connect stations B and C, as shown in Figure 6.

Figure 4
Connection during NXFER

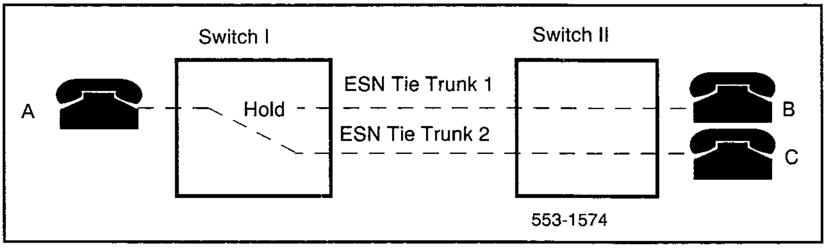


Figure 5
Connection after NXFER

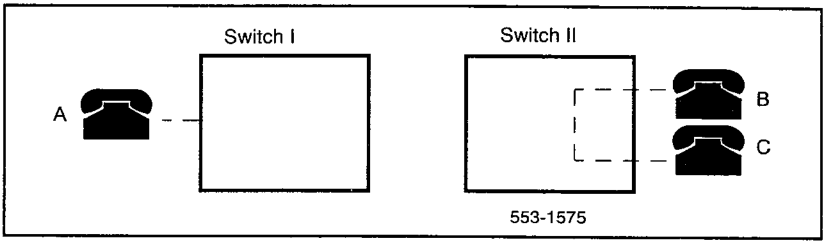
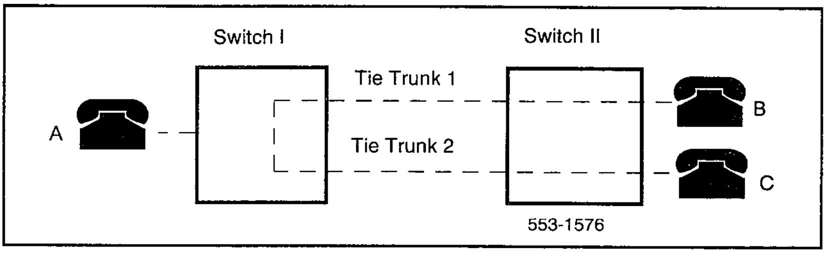


Figure 6
Connection after XFER



1+ dialing

With 1+ dialing, both NARS translation tables are equipped for four-digit translation (based on the first one to four digits), thus allowing NARS customers the option of dialing 1+ after the NARS access code for long distance calls. See Table 3 for dialing formats for NARS Uniform Dialing Plan (UDP) calls.

The 1+ dialing capability also eliminates ambiguity between identical three-digit NPA, NXX, and LOC codes so that the NARS customer can route calls to an NPA, NXX, or LOC code that conflicts with one of the customer's three-digit codes.

If 1+ dialing is specified for an NPA, NXX, or SPN number in a translator, the digit "1" must not be used as the leading digit for Network Speed Call list codes in that translator.

Network Control

Network Control (NCTL) is an enhancement to the NCOS feature that extends NCOS controls to users located at ESN main switches. Network Control requires that the ESN main and serving ESN node be equipped with the Network Signaling (NSIG) feature.

ESN main NCOS

Users (lines, trunks, attendants) at an ESN main are assigned an NCOS, which is used to determine their level of access to network facilities at the serving ESN node. When a user at an ESN main initiates a call to, or through, an ESN node, the user's assigned NCOS can be transmitted on Traveling Class of Service (TCOS), depending on tie trunk settings (ESN, ESN2, etc.). If the user's NCOS is transmitted to the serving node and the node is equipped with Basic Alternate Route Selection (BARS), only NCOS 0 to 7 can be assigned at the node. Therefore, only NCOS 0 to 7 should be assigned at the ESN main even though an ESN main with NSIG can have NCOS 0 to 15.

If the node is equipped with NARS, then NCOS 0 to 15 can be assigned both at the node and at the ESN main. The transmitted NCOS (or TCOS) overrides the NCOS (or FRL) assigned to the incoming tie trunk group at the node, and it is used to determine the user's eligibility for network resources/features at the ESN node. Thus, a user at an ESN main has the same network access capabilities as a user at the ESN node who is assigned the same NCOS.

> *Note:* If the user at the ESN main enters a valid authorization code prior to placing a NARS call, the NCOS associated with the authorization code is transmitted to the ESN node in place of the user's assigned NCOS.

Calls from a Conventional main to the ESN node are controlled by the NCOS assigned to the incoming trunk group at the ESN node, since the Conventional main does not have NSIG.

ESN node TCOS

Network Control (NCTL) at an ESN node can provide a Traveling Class of Service (TCOS) mechanism that controls route access and Off-Hook Queuing (OHQ) eligibility for calls placed to or through another ESN node or an associated ESN main, it enables the ESN node to interface with switches that are part of an Electronic Tandem Network (ETN), provided that the SIGO setting on the tie trunk is set for ETN at both ends.

The Traveling Class of Service (TCOS) is, in effect, the FRL of a user's assigned NCOS. When a user at an ESN node initiates a call to another ESN node or an ESN main, the TCOS (for example, the FRL of the user's assigned NCOS) is transmitted to the other ESN node. At the receiving ESN node, the TCOS (0 to 7) replaces the FRL of the NCOS assigned to the incoming trunk group.

Route access and OHQ eligibility for the call are, therefore, based on the NCOS of the incoming trunk group with the modified FRL (for example, TCOS).

> *Note:* The Network Control (NCTL) data block (LD87) is used to define OHQ eligibility on a per FRL (TCOS) basis. For example, if FRL 4 is defined as OHQ eligible, then all users who have an NCOS with a FRL of 4 are eligible for OHQ on calls placed to another ESN node or to an associated ESN main.

If a user at an ESN main or Conventional main initiates a call that tandems through the serving ESN node to another ESN node or ESN main, TCOS applies to the call as if the call originated at the serving ESN node.

ETN switch compatibility

The TCOS is equivalent to the Traveling Class Mark (TCM) used at Electronic Tandem Network (ETN) switches. See Technical Publication 42709, "Tie Trunk Signaling Compatibility for Connecting to a DIMENSION PBX," July 1979.

When a ten-digit UDP call or Distance Steering Code (DSC) Coordinated Dialing Plan (CDP) call is made from an ESN node to an Electronic Tandem Network (ETN) switch, the dialed digits, together with the TCOS number (0 to 7), are sent to the connected ETN switch. At the ETN switch, the TCOS number received from the ESN node is used as a Traveling Class Mark (TCM) to determine route access and OHQ eligibility at the ETN switch.

Similarly, when a call is made from an ETN switch to an ESN node, the dialed digits, together with the Traveling Class Mark (TCM) number (0 to 7), are sent to the connected ESN node. The ESN node interprets the received TCM number as a TCOS number. The received TCM (for example, TCOS) replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch. This new FRL (for example, TCM) is then used to determine route access and OHQ eligibility for the call.

However, if a Distance Steering Code (DSC) CDP call is terminated on a switch as a Local Steering Code (LSC) call, the TCOS value transmitted by the connected switch will not be collected and saved by this switch.

Network Signaling

The Network Signaling (NSIG) feature provides the required signaling protocol to interface nodes with ESN mains, nodes with other nodes, and nodes with Electronic Tie Network (ETN) switches.

Installing NSIG at a Conventional main enhances it and it becomes an ESN main. When callers at an ESN main place calls through a node with NSIG, their NCOS or TCOS travel with the call and are interpreted at other NSIG equipped switches. The tie trunk settings determine and control the operation of this feature.

When the NSIG feature is equipped at a switch, there are options available (Route Data Block [RDB], LD16) to define the signaling arrangements between that switch and any other switch connected to it by means of tie trunks. These options define what call information is to be transmitted to a connected switch and what call information is to be received from a connected switch. The option selected depends on the type of connected switch (ESN node, ESN main, Conventional main, ETN) and the options (for example, Coordinated Call-Back Queuing [CCBQ], Call-Back Queuing to Conventional Mains [CBQCM]) that are available to the connected switch.

The signaling options are STD (standard), ESN, ESN2, ESN3, ESN5 (Electronic Switched Network), and ETN (Electronic Tie Network).

STD Arranges the tie trunk group for transmission/reception of the called number between switches. Sends outpulsed digits.

ESN (X11 release 2 only) Arranges the tie trunk group for transmission/reception of the call type, NCOS/TCOS, and called number between switches, and is required on systems equipped with the CCBQ/CBQCM feature. Sends call type, NCOS or TCOS, and dialed digits.

ESN2 (X11 release 3 and later) Arranges the tie group as described for ESN in X11 release 2. Used unless switch has NXFER or Satellite Link Control (SAT).

ESN3 (X11 release 3) Arranges the tie group as described for ESN in X11 release 2 and is required on systems equipped with the NXFER or Satellite Link Control features.

ESN5 (X11 release 5 and later) Arranges the tie group as described for ESN in X11 release 2; needed with Digital Trunk Interface (DTI).

ETN Arranges the tie trunk group for transmission/reception of the called number and TCOS/TCM between switches and is used when connected to an ETN switch. Sends outpulsed digits and TCOS.

Application

Following is a description of how these options are applied to accommodate the different switch types that can be connected to an ESN main or ESN node that is equipped with the NSIG feature.

ESN node An ESN node can be connected by means of tie trunks to another ESN node, an ESN main, a Conventional main, and/or an ETN Switch.

- If the ESN node connects to another ESN node, both ends of the connecting tie trunk group are defined with the ETN option.
- If the ESN node is equipped with X11 release 2 and connects to an ESN main equipped with X11 release 2, both ends of the connecting tie trunk group are defined with the ESN option.
- If the ESN node is equipped with X11 release 3 and the ESN main is equipped with X11 release 3, both ends of the connecting tie trunk group are defined with the ESN3 option.
- If the ESN node is equipped with X11 release 2 and the ESN main is equipped with X11 release 3, the node end of the connecting tie trunk group is defined with the ESN option, and the main end is defined with the ESN2 option.
- If the ESN node is equipped with X11 release 3 and the ESN main is equipped with X11 release 2, the node end of the connecting tie trunk group is defined with the ESN2 option, and the main end is defined with the ESN option.

- If the ESN node connects to a Conventional main, the node-end of the tie trunk group is defined with the STD option.
- If the ESN node connects to an ETN switch, the node-end of the tie trunk group is defined with the ETN option.

ESN main An ESN main can be connected by means of tie trunks to an ESN node and satellite switches.

- If the ESN main is equipped with X11 release 2 and connects to an node equipped with X11 release 2, both ends of the connecting tie trunk group are defined with the ESN option.
- If the ESN main is equipped with X11 release 3 and the ESN node is equipped with X11 release 3, both ends of the connecting tie trunk group are defined with the ESN3 option.
- If the ESN main is equipped with X11 release 2 and the ESN node is equipped with X11 release 3, the Main end of the connecting tie trunk group is defined with the ESN option, and the node end is defined with the ESN2 option.
- If the ESN main is equipped with X11 release 3 and the ESN node is equipped with X11 release 2, the Main end of the connecting tie trunk group is defined with the ESN2 option, and the node end is defined with the ESN option.
- If there are satellite switches (non-ESN) connected to the ESN main, the main-end of the tie trunk groups from the satellite switches are defined with the Standard (STD) option.

Requirements

The following requirements apply:

- An ESN main can connect to only one ESN node. Both switches must have NSIG for NSIG related features.
- Tie trunks between ESN nodes and ESN mains must be arranged for Dual Tone Multi-Frequency (DTMF) sending/receiving, wink-start operation and must have answer supervision.
- ESN node compatibility with ETN switches is limited to seven-digit on-network, ten-digit off-network, and Distance Steering Code (DSC) CDP calls.

Satellite Link Control

Tandem trunk calls, when connected through more than one communications satellite trunk, are subject to transmission distortion due to propagation to and from communications satellites. The Satellite Link Control (SAT) feature ensures that the configuration of a call does not include more than one communications satellite trunk.

Requirements

- This feature applies to ESN network calls (NARS/BARS/CDP) only.
- ESN Proprietary Signaling (NSIG) is required among ESN switches.
- Routes that receive digits from satellites or send digits to satellites have to be marked as SATELLITE routes.

Routing Control

The Routing Control (RTC) feature provides a mechanism for changing a user's network-access capabilities when a special TOD schedule is in effect, when an extended TOD is in effect, or when the user presses a Routing Control key on the console.

NCOS map

With the NARS feature, TOD schedule 7 is the special TOD schedule. Associated with the special TOD schedule is a NCOS map. The NCOS map lists all NCOS numbers.

Associated with each listed NCOS is an alternate NCOS number (greater than, equal to, or smaller than) that replaces the original NCOS number when the special TOD schedule is in effect. Table 7 illustrates a typical NCOS map.

Table 7
Typical NCOS map for special TOD schedule 7

Original NCOS	Alternate NCOS	Original NCOS	Alternate NCOS
0	0	8	2
1	0	9	3
2	0	10	3
3	1	11	4
4	1	12	4
5	2	13	5
6	2	14	5
7	2	15	5

Note: When TOD Schedule 7 is in effect, the alternate NCOS replaces the user's original NCOS.

Invoking Routing Control

The alternate NCOS numbers associated with special TOD schedule 7 are normally invoked when the time specified for TOD schedule 7 corresponds to the time in the system clock. Additionally, the alternate NCOS numbers can be scheduled for implementation (through service change) for the full 24-hour period of specified days of the week. This capability enables network-access capabilities to be reduced automatically on weekends or company holidays.

The attendant can also manually invoke the special TOD schedule by using a Routing Control (RTC) key on the console. Pressing the RTC key lights the associated lamp and invokes the special TOD schedule. To deactivate Routing Control, the RTC key is pressed again. The associated lamp goes dark, and normal TOD schedules are once again in effect.

> *Note:* Authorization Code can be used to override the restrictions imposed through routing control. If a user enters a valid authcode, the NCOS number associated with the authcode is applied for the duration of the call.

Call-Back Queuing

Call-Back Queuing (CBQ) is an ESN node feature that provides queuing for network calls encountering busy or blocked facilities at the ESN node. CBQ enables the calling party to go on hook after activation of the Ring Again (RGA) feature and receive a call back from the ESN node when a network facility becomes available. See *X11 features and services* (553-3001-305).

The CBQ feature is available only to stations located at an ESN node. Unlike OHQ, CBQ is offered only at the originating ESN node. Access to CBQ is accomplished through the existing Ring Again (RGA) feature.

Options

Two options for CBQ eligibility are defined by the call originator's NCOS. The first option, CBQ(i), means that CBQ may be offered after only the initial route set of a route list has been examined for an available route. The second option, CBQ(a), means that CBQ may be offered after both the initial and extended (for example, all) route sets of a route list have been examined. In either case, a call offered CBQ is first queued against the initial route set.

Eligibility

Before offering CBQ to a call originator, the following eligibility tests are performed.

- At least one of the routes in the initial route set is defined as CBQ eligible.
- The user's NCOS is defined as permitting CBQ; either CBQ(i) or CBQ(a).
- The call is not eligible for OHQ. Calls that are eligible for both OHQ and CBQ will be offered OHQ.
- The user's telephone is allowed access to the Ring Again feature and does not have another CBQ or Ring Again call already in the queue.
- The CBQ feature is enabled for this customer.

CBQ(i) eligible For call originations by a caller defined as CBQ(i) eligible, the system searches the initial route set for an available route. If no available route is found, CBQ is offered to the caller subject to the CBQ eligibility tests.

CBQ(a) eligible For call originations by a caller defined as CBQ(a) eligible, the system examines the initial route set for an available route. If no available route is found, the extended route set is then searched for an available route. If an available route is not found in the extended set, then CBQ is offered, subject to the CBQ eligibility tests. However, if an available route in the extended route set is found that is designated as expensive, the user's NCOS allows ERWT, and ERWT is enabled for the customer, the tone is given and the system delays terminating the call. During this delay the user has the following options:

- Refuse the expensive route by abandoning the call.
- Wait and allow the call to complete over the expensive route.
- Activate the Ring Again feature (feature key or access code) to place the call in the CBQ. The user must be CBQ(a) eligible; otherwise, operation of the Ring Again feature is ignored.

Offer

The CBQ offer consists of an optional recorded announcement, followed by overflow tone. If the user wishes to accept the CBQ offer, Ring Again must be activated within 30 seconds. Ring Again activation follows present feature operation for SL-1 and 500/2500 telephones. See *X11 features and services* (553-3001-305). The CBQ offer can be refused by going on hook. If the user neither accepts nor refuses the CBQ offer within 30 seconds, the call is force disconnected. If the user wishes to accept the CBQ offer, Ring Again must be activated within the programmed time (default is 20 seconds).

CBQ calls are placed in a priority-ordered trunk queue (together with OHQ calls, if any) with a starting priority and maximum priority (see LD87) defined by the call originator's NCOS. (Refer to "Off-Hook Queuing" on page 51 for additional information on priority queuing.) At the same time, two timers are started.

- a queue promotion timer and a route advance timer
- each with values defined through the originator's NCOS

At intervals defined by the queue promotion timer, the priority of the call is incremented until it reaches its maximum priority.

Each time the call priority is incremented, its position in the CBQ is advanced. If the route advance timer reaches its maximum value before the call can be terminated on a route in the initial set (I set), the extended set of routes is added to the routes that the call is currently queued against.

> *Note:* A route advance timer (RADT) set to "0" never expires. The user always queues for I set routes only.

Expensive route warning tone is not given to calls that have been queued, even if terminated on expensive facilities. Unless cancelled by the call originator, CBQ calls remain in the queue until they have been offered a trunk; there is no time limit on CBQ calls. Calls can only be routed on routes in the I set or extended set if the FRL in the NCOS is equal to or greater than the FRL assigned to the route in the route list.

Call back

When a trunk becomes available for a CBQ call, it is seized to prevent incoming originations during the CBQ call back period. Outpulsing of digits (either those originally dialed by the user or those required as a result of digit manipulation) is started at a slow, fixed rate. The number of digits to be outpulsed determines how long the trunk can be held while the CBQ call back is being offered to the originating station. The system computes this time by allowing 10 seconds before the first digit is outpulsed and 256 seconds between subsequent digits.

The originator of the call is alerted to the CBQ call back by either tone buzzing and a winking Ring Again feature lamp (SL-1 telephone), or short bursts of ringing (500/2500 telephone). The SL-1 telephone user must accept the call back within the computed value of outpulse time or the service-changeable CBQ time limit of 10 to 30 seconds, whichever is less. A user with a 500/2500 telephone must accept the call back within 6 seconds. Acceptance of the CBQ call back is performed with present Ring Again operating procedures.

When a CBQ call back is answered at a digit display SL-1 telephone, the originally dialed digits are displayed. If the user does not answer the call back within the time limit, the call is removed from the queue and discarded. If the user accepts the call back within the time limit, the call is terminated. A CBQ call can be canceled by the originating station by means of the existing procedures for Ring Again cancellation.

Feature interactions

The CBQ feature interacts with the following features.

- **Barge-In, Force Disconnect** Between the time a trunk is seized for a CBQ call and the user accepts the CBQ call back, the trunk can be stolen by the attendant or force disconnected through service change. If this occurs, there is no guarantee that the call can be terminated when the user accepts the CBQ callback. Under these circumstances, the call is treated like a new origination and NARS/BARS is used to reattempt termination. This can result in the call being blocked and being offered CBQ a second time.
- **Hunting, Call Forward, Multiple Appearance DN** CBQ call backs to stations at an ESN node are offered only to the originating station, regardless of the hunting or call forwarding that may be in effect. Other appearances of a station's Directory Number (DN) are not offered the call back.
- **Attendant functions** Because the Ring Again feature is not supported at attendant consoles, CBQ is not offered to an attendant regardless of the CBQ eligibility of the NCOS assigned to the attendant.

Off-Hook Queuing

Off-Hook Queuing (OHQ) is a software feature that can be equipped at an ESN node and/or ESN main. The feature enables a call originator to remain off hook for a short time (customer programmable) until a network facility for call completion becomes available at the ESN node or ESN main.

Eligibility

Network calls may be placed in an OHQ if all trunk routes (entries) in the initial route set of a route list (see note) are busy, and the following criteria are met:

- OHQ has been allowed for that customer group.
- At least one of the trunk routes in the initial route set of a route list is defined as being eligible for OHQ.
- The NCOS of the call originator (at an ESN node or an ESN main) is defined to permit OHQ.
- The incoming trunk group at the ESN node or ESN main is defined in software to permit OHQ for incoming calls.
- The TCM received at the ESN node from an ETN switch is compatible with a FRL that is defined to permit OHQ.
- The TCOS received at the ESN node from another ESN node is compatible with an FRL that is defined to permit OHQ.
- The probability that the call times out in the OHQ before a trunk becomes available is below a specific threshold.
- The OHQ feature is enabled.

Note: A marker determines which route list entries are in the initial route set (see automatic least-cost routing). Typically, the initial route set contains the cheaper routes to a particular destination. The remaining routes in the route list, if any, comprise the extended route set and are usually the more expensive routes to the destination. Only routes in the initial route set should be defined to allow OHQ. OHQ are not offered by routes in the extended route set even if they are defined to allow OHQ.

Calls that do not meet the preceding requirements for OHQ eligibility can be offered CBQ at this point.

Availability

The OHQ software performs an "availability" test to prevent calls from entering the OHQ when there is a high probability that the call will time out before a trunk becomes available. The following process is used to make this test:

- For each trunk route, a counter is maintained that reflects the current number of calls with the maximum queue priority of 3 queued against it. This includes all calls in OHQ, Ring Again, and those CBQ calls that are currently at priority 3 as well as any direct-access calls in progress.
- Each trunk route has a threshold value that indicates the maximum number of priority 3 calls that can be queued against it before OHQ timeout becomes a high probability. Before a call is placed in the OHQ, the current queue count is compared with the threshold value for each eligible trunk route in the initial set of routes. If at least one of the trunk routes has a count that is less than or equal to the threshold value, the call is allowed to OHQ against all OHQ eligible routes.

  Note: A maximum priority (0, 1, 2, or 3) and a starting priority (0, 1, 2, or 3) are assigned to each user through LD87. Zero is the lowest priority while three is the highest. Calls are queued according to their starting priority and move to a higher queue (up to their maximum priority) as their promotion timer allows. Once calls reach their maximum priority, they wait for an available trunk in the I set. If the route advance timer (RADT) expires, they can also queue for the extended set routes.

Offer

If all eligibility and availability checks are successful, the call originator is given a distinctive OHQ offer tone (a 1 second burst of 440-Hz tone) when the call enters the OHQ. The call originator either accepts the OHQ offer by remaining off hook or declines the offer by going on hook.

OHQ calls are placed in a priority-ordered queue with all other calls waiting for trunks. OHQ calls are assigned the maximum priority (3), since other network facilities may be held while the call is queued. A timer with an initial value equal to the software-defined OHQ time limit is started to limit the duration of the OHQ. The OHQ time limit is service changeable within the range of 2 to 60 seconds.

The queue is examined whenever a trunk becomes idle in a trunk route on which one or more calls are queued. If a call is found that can be terminated on an idle trunk, the available trunk is seized and the call terminated.

If the OHQ timer expires before the call can be terminated, the call is removed from the OHQ. At this time, the remaining eligible routes in the route list (the extended set) are examined, and the call is either terminated or given standard blocking treatment if no facilities are available. CBQ will not be offered at this point as OHQ was already offered. The ERWT is not given to calls that have been queued, even if terminated on expensive facilities.

OHQ can be offered to call originators located at an ESN node, ESN main, Conventional main or ETN switch. Also, as a call progresses through the network, OHQ can be offered to the call originator from any of the ESN nodes or ESN mains that are used to process the call. For example, OHQ can be offered more than once for a given call.

Feature interactions

The OHQ feature has the following interactions with existing features:

Call modification

Station users are not allowed to activate call modification features (hold, call transfer, conference) while waiting in the OHQ. Switchhook flash used to activate features from 500/2500 telephones is ignored. Similarly, operation of set feature keys is ignored.

Camp-On, Call Waiting

If the attendant extends a call to a telephone that is in the OHQ, the Call-Waiting tone is not offered to the telephone. If the attendant releases, the call is camped on the OHQ telephone, but no warning tone is given. When the Camp-On call is recalled to the attendant console, the attendant can repeat the Camp-On procedure. Once the OHQ call is in an established state, the Camp-On tone is provided.

Attendant functions

- The attendant cannot Barge-In during trunk seizure for OHQ calls.
- If the attendant extends a network call for a telephone user and the call is offered OHQ, the attendant must inform the caller of the OHQ offer before releasing from the call.
- The attendant is not allowed to operate the Release key or another Loop Key (LPK) if the source call is in conference and the destination call is in the OHQ. Operation of the Release Destination key is permitted, however, and causes the OHQ call to be abandoned.

Coordinated Call-Back Queuing

The Coordinated Call-Back Queuing (CCBQ) feature enables stations at an ESN main to be offered CBQ when network calls are blocked at the serving ESN node. When facilities become available at the ESN node, the call originator at the ESN main is alerted by a call back from the node. This feature requires that the ESN main and associated ESN node be equipped with the Network Signaling (NSIG) feature.

Eligibility

When a telephone at an ESN main originates a network call through an ESN node, the NCOS of the call originator, the call type, and whether or not the telephone is allowed access to the Ring Again feature is transmitted to the ESN node. If an authcode is entered at the ESN main prior to dialing a network call, the NCOS associated with the authcode is transmitted to the ESN node. When received by the node, this NCOS is used to determine CCBQ eligibility and is used for the duration of the call, unless further modified by the Authcode Conditionally Last feature.

The CBQ eligibility tests are performed. In addition, a check is made that the incoming trunk group from the ESN main is defined (at the ESN node) to permit CBQ and that the call type allows CBQ. CCBQ is offered to the user at the ESN main if the eligibility tests are successful. If the tests are unsuccessful, standard call blocking is applied to the call.

As for stations at an ESN node, the call originator at an ESN main can invoke Ring Again upon receipt of ERWT if the originator's NCOS is defined at the ESN main as CBQ(a) eligible.

Offer

The CCBQ offer and acceptance sequences are identical to those for stations at the ESN node (see "Call-Back Queuing" on page 45). The optional recorded announcement and overflow tone are provided by the ESN node. In addition, after the recorded announcement is provided, the ESN node transmits a signal to the ESN main. This signal indicates that the call is in a state that allows Ring Again.

When the call originator at the ESN main activates Ring Again, the ESN main assigns a unique "queue identification" number to the call. This number is transmitted to the ESN node to indicate CCBQ acceptance. At the ESN main, the call is placed in a holding queue. At the ESN node, the call, together with the queue identification number, is placed into the trunk queue. The ESN main to ESN node tie trunk is released.

Call back

When an outgoing trunk is seized by the ESN node for a CCBQ call, slow outpulsing is started to hold the trunk while a call back is made to the call originator at the ESN main. The ESN node seizes an available tie trunk to the ESN main and transmits the "queue identification" number of the call to the ESN main. The ESN main then initiates a call back to the call originator. Callback presentation to the call originator is as for standard Ring Again (see "Call-Back Queuing" on page 45).

> *Note:* If no tie trunks to the ESN main are idle, the outgoing trunk is released and can be offered to another call. The CCBQ call retains its position in the queue but is not offered another trunk until a tie trunk to the ESN main becomes available.

When the call originator at the ESN main accepts the CCBQ call back, answer supervision is sent from the ESN main to the ESN node. The ESN node completes the call, and the digits are outpulsed at a normal rate.

If the call originator is equipped with a 500/2500 telephone and is engaged in a call when the ESN node initiates a CCBQ call back, a signal is transmitted from the ESN main to the ESN node. The ESN node releases the outgoing trunk and places the CCBQ call into a holding queue for 5 minutes. No attempt is made to seize another outgoing trunk for the call until the holding time expires. This process occurs only once.

If the originating telephone is still busy, the CCBQ is canceled automatically at the ESN node. No indication is given to the call originator of the CCBQ cancellation. To prevent the CCBQ call from remaining indefinitely in the holding queue at the ESN main, the ESN main sets a time limit of 1 hour for CCBQ calls. When this time limit expires, the CCBQ call is canceled automatically. CCBQ call back to a busy telephone is as for normal Ring Again.

The call originator at the ESN main can cancel the CCBQ call at any time; however, the ESN node is not aware of the cancellation until the CCBQ call back is attempted.

Feature interactions

The CCBQ feature interacts with the following features:

- **Initialize** If the main initializes while calls are queued at the node, CCBQ call backs from the node are not answered because the initialization has cleared the holding queue at the main. The node treats these calls as call-back-no-answer calls and cancels the CCBQ automatically. If the node initializes, CCBQ calls that are in the trunk queue are lost. The main cannot detect this situation. To prevent calls from remaining indefinitely in the holding queue at the main, the main sets a time limit of 1 hour for CCBQ calls. If a call back from the node is not received within 1 hour, the main cancels the CCBQ call automatically.
- **Attendant functions** Attendants at an main are not offered CCBQ. Attendant Barge-In on trunks involved in CCBQ operations results in cancellation of the CCBQ call.
- **AIOD and ANI** Automatic Identification of Outward Dial (AIOD) and Automatic Number Identification (ANI) facilities can be used to complete CCBQ calls from an ESN node. The outgoing toll call is billed to the access tie trunk rather than the telephone at the ESN main.

Coordinated Call-Back Queuing Against Main

The Coordinated Call-Back Queuing Against Main (CCBQAM) feature enables stations at nodes to be offered CBQ for network calls that are blocked at a main. When facilities become available at the main, the call originator at the node is alerted by a call back from the main. CCBQAM otherwise functions identically to CCBQ at the node.

Call-Back Queuing to Conventional Mains

The Call-Back Queuing to Conventional Mains (CBQCM) feature allows call originators at a Conventional main to access the CBQ feature at an ESN node.

Eligibility

When a telephone at a Conventional main originates a network call through an ESN node, the NCOS assigned to the incoming trunk group is used to determine the Call-Back Queuing to Conventional Mains (CBQCM) eligibility. This NCOS, as well as the incoming trunk group, must be defined as CBQ eligible.

Offer

The Call-Back Queuing to Conventional Mains (CBQCM) offer to the call originator at a Conventional main consists of an optional recorded announcement followed by special (interrupted) dial tone. (The announcement and tones are provided from the ESN node.) To accept the CBQCM offer, the call originator dials the extension number associated with the telephone being used for the call. When the last digit of the extension number is dialed, a confirmation tone (three 256-ms bursts of dial tone) is sent from the ESN node to the call originator. The call is placed in the CBQ at the ESN node when the call originator goes on hook.

The CBQCM offer can be refused by going on hook any time before the last digit of the extension number is dialed, or by remaining off hook for longer than 30 seconds after receipt of the confirmation tone. If the CBQCM is neither accepted nor rejected within 30 seconds, the caller is given overflow tone (from the ESN node) and the call is disconnected.

Call back

When an outgoing trunk becomes available at the ESN node, it is seized and slow outpulsing is started. The ESN node then seizes a tie trunk (see note) to the Conventional main and outpulses the extension number of the call originator. The call originator must answer the call back before slow outpulsing is completed; otherwise, the call back is canceled and the outgoing trunk is released.

> *Note:* If no tie trunks are currently available to the Conventional main, the node releases the outgoing trunk. The CBQCM call retains its position in the queue but is not offered another outgoing trunk until a tie trunk to the Conventional main becomes available.

When the call originator answers the CBQCM call back, answer supervision must be transmitted from the Conventional main to the ESN node. Upon receipt of answer supervision from the Conventional main, the ESN node transmits a tone (three 256-ms bursts of dial tone) to notify the call originator that the call is a CBQCM call back, and completes the call.

If the call originator's telephone is busy, or the originator does not answer when the call back is placed, the ESN node places the call in a suspended state for 5 minutes. After 5 minutes, another call back is attempted if the outgoing trunk is free. If the telephone that originated the call is still busy or does not answer, the ESN node cancels the call back.

No provision is made for CBQCM cancellation by a call originator at a Conventional main. Once the CBQCM offer is accepted, the call remains in the queue until the ESN node initiates a call back.

Requirements

Station users at Conventional Mains cannot activate Ring Again to refuse expensive routes after the ERWT is given.

The ESN node seizes the same tie trunk group that was used to initiate CBQCM for the CBQCM call back. Thus, these trunk groups must be two-way (incoming and outgoing).

Conventional mains must provide answer supervision on tie trunks connected to the ESN node. These switches must also permit transmission or repetition of telephone dial pulses for CBQCM operation. This feature cannot be used with systems that operate in senderized mode. Operation may require adjustment of the interdigit timeout on systems that employ simulated cut-through operation.

Multiple call back queues are allowed per trunk group for the Conventional main by dialing any digits (up to 7) based on the availability of SL-1 call registers.

Conventional mains must not allow CBQCM call back calls to be modified by call transfer or call forward to an outside line. Call modifications like this can result in the tie trunk not being released at the end of the call.

Network Authorization Codes

The Authorization Code feature enables selected users to temporarily override the access restrictions assigned to a station or trunk. A user can enter an authorization code (authcode) to access more of the system facilities than would normally be allowed to the particular station or trunk because of the assigned Network Class of Service (NCOS), Class of Service (CLS), and Trunk Group Access Restriction (TGAR) codes.

This feature is useful when a user initiates a call from someone else's telephone and requires access to more system facilities, for example, access to long distance calling, than are allowed to that telephone. Entering a valid authcode enables the user to access these additional features. Once a valid authcode is entered, the NCOS, CLS, and TGAR associated with the authcode replace the NCOS, CLS, and TGAR associated with the telephone for the duration of the call.

The Network Authorization Code (NAUT) feature provides the customer with two package options:

— Basic Authorization Codes (BAUT) for general applications

— Network Authorization Codes (NAUT) for network applications

Basic Authorization Codes

The Basic Authorization Code (BAUT) package provides for up to 4096 authcodes of 1 to 14 digits. Users can enter an authcode after dialing the Special Service Prefix (SSP) and the digit "6," before dialing any call, including a NARS call. With the BAUT package, an authcode can be entered when:

- originating a call from a local station or tie trunk
- initiating a call transfer or conference from a local station
- originating a call by means of the Direct Inward System Access (DISA) feature

Network Authorization Codes

The Network Authorization Codes (NAUT) package provides for up to 20,000 authcodes of 1 to 7 digits. With X11 release 13 and later, the authcodes can have from 1 to 14 digits. The NAUT package incorporates all the features of the Basic Authorization Code (BAUT) package, adds a "conditionally last" option for entering an authcode after dialing a NARS call, and allows the attendant to enter an authcode.

Authcode Conditionally Last

With the NAUT package, users can be prompted "conditionally" for an authcode after dialing a NARS call. The prompt is by an "authcode request," which consists of 10 bursts of dial tone, followed by steady dial tone. The authcode request can optionally be preceded with an appropriate recorded announcement. The user is prompted for an authcode entry only if:

- an authcode was not previously entered
- the FRL associated with the user's NCOS is less than the service change–assigned minimum FRL of the route list that NARS would use for the call.

Users at an ESN main or Conventional main connected by means of tie trunks to an ESN node can optionally be prompted for an authcode entry after dialing a NARS call. The user is prompted for an authcode entry only if:

– no authcode was previously entered

– the FRL associated with the NCOS of the incoming (or two-way) tie trunk, or the caller's NCOS if it is being sent, is less than the minimum FRL of the route list that NARS would use for the call

– the route is defined in the Route Data Block (RDB), LD16, to prompt for an authcode entry on incoming NARS calls.

Users accessing an ESN node by means of the Direct Inward System Access (DISA) feature to make a BARS/NARS call are prompted for an authcode entry if:

– no authcode was previously entered

– the FRL of the NCOS assigned to the DISA Directory Number (DN) is less than the minimum FRL of the route list that NARS would use for the call

Attendant input of authcode

Normally, because an attendant is not restricted from accessing any system resource, the attendant does not need to have an authcode. The Network Authorization Code package enables the attendant to enter an authcode for other callers; for example, the attendant can enter an authcode, after dialing the Special Service Prefix (SSP) and the digit "6," and complete a long distance call for a local user whose CLS is Toll Denied (TLD). If the Call Detail Recording (CDR) of authcodes is defined for the customer, the local user's authcode digits appear in the CDR for billing purposes.

Attendants are normally assigned an NCOS having a high FRL so that they can make any type of call, including NARS calls. An attendant can be prompted for an authcode entry if the FRL required to access a route list for a NARS call is greater than the FRL of the attendant's NCOS.

Authcode validation

The software validates an inputted authcode on the basis of the number of digits dialed and the dialed digits themselves. If the authcode contains more or fewer digits than the defined authcode length (Authcode Data Block [AUB], LD88), the authcode is invalid. Similarly, if the dialed authcode digits are not defined in the Authcode Table (AUT), LD88, the authcode is invalid. When an invalid authcode is encountered, no response is given to the user until the End-of-Dialing (EOD) timer expires. This increases the security of authcodes by making it difficult for an unauthorized user to determine the length of a valid authcode. When the EOD timer expires, overflow tone is given for 15 seconds and the call is force disconnected.

Authcode administration

With the NAUT and BAUT packages, a "classcode" structure is used as part of authcode administration. A classcode is a definition of a combination of CLS, Trunk Group Access Restrictions (TGAR), and NCOS codes. There can be up to 116 (0 to 115) classcodes defined through the Authcode Data Block (AUB), LD88, each having a different combination of CLS, TGAR, and NCOS codes. Each authcode is associated with a classcode. Authcodes that have the same combination of CLS, TGAR, and NCOS codes are assigned the same classcode.

With the NAUT package, authcodes can be defined individually by the customer or generated automatically by the system. When defining or generating new authcodes, a classcode with which the new authcodes are to be associated is specified. The new authcode is then automatically assigned the CLS, TGAR, and NCOS codes associated with the specified classcode.

When an authcode is to be removed from use, a facility exists to prevent that authcode from being reused. For example, the authcode is not accepted as valid input when individually defining authcodes. This is accomplished through an "exemptcode."

When an authcode is removed from use, an exemptcode is assigned to the authcode in place of the classcode. The exemptcode is the month, for example, JAN, FEB, taken from the system clock. If an exemptcode is not requested, the removed authcode is returned to the pool of unused authcodes and can be reused at any time.

The Route List Block (RLB) program (LD86) is used to define a minimum FRL for each route list. This minimum FRL (range 0 to 7) is used to determine whether or not to prompt for an authcode entry after a call. If a minimum FRL is not specified, the actual minimum FRL in the initial route set is used. Similarly, the Route Data Block (RDB) program (LD16) is used to define whether or not to prompt for an authcode entry on calls on incoming or two-way tie trunk groups.

Feature operation

Use the authcode after SSP (500/2500/SL-1 telephones). To enter an authcode after the Special Service Prefix (SSP), the caller proceeds as follows:

1 If a call is not in progress, go off hook or press a DN key. If there is a call in progress, switchhook flash (500/2500 telephone) or press the Call Transfer or Conference key (SL-1 telephone) to obtain special dial tone.

2 Dial the authcode access number (SSP and the digit "6"). Dial tone is removed after the SSP digit is dialed.

3 Dial the authcode digits. A second dial tone sounds if the authcode is valid. If the authcode is invalid, no response is given until the End-of-Dialing (EOD) timer expires, then overflow tone is given for 15 seconds and the call is force disconnected.

4 When the second dial tone sounds, dial the call in the normal manner. If Call Transfer/Conference is in effect, complete the transfer/conference as normal.

Authcode after SSP (attendant)

To enter an authcode after SSP, the attendant proceeds as follows:

1 If there is a call on the source loop, go to step 2. If there is no call on the source loop, press an idle Loop key (LPK).

2 Dial the authcode access number (SSP and the digit "6"), followed by the authcode.

3 Dial as usual after receiving dial tone denoting a valid authorization code. If the code is invalid, overflow tone is returned immediately.

Authcode Conditionally Last

The following procedure is used to enter an Authcode Conditionally Last from a 500/2500/SL-1 telephone or attendant console (NAUT package only):

1 Place a call.

2 Receive an "authcode request" (optional recorded announcement followed by 10 bursts of dial tone, followed by steady dial tone), indicating that an authcode entry is required.

3 Dial the authcode. Dial tone is removed after the first digit is dialed. If the authcode is valid, the call is processed as a normal call. If the authcode is invalid, overflow tone is returned when the End-of-Dialing (EOD) timer expires.

Feature interactions

Feature key operations While a user is entering an authcode, the following feature keys operate as intended and do not affect operation of the Authorization Code feature:

- Make Set Busy
- Buzz
- Volume Control

The following key operations are ignored during authcode operation:

- Conference
- Override
- Call Forward
- Call Transfer
- Call Pickup
- Charge Account
- Calling Party Number
- Privacy Release
- Ring Again
- Barge-In and Busy Verify

- Speed Call
- Recall
- Do Not Disturb
- Digit Display

The following key operations abort the authcode operation, and any digits entered for an authcode are ignored:

- Directory Number
- Paging
- Voice Call
- Not Ready
- In-Calls
- Call Waiting
- Hold
- Release

If the caller initiates a switchhook flash while entering an authcode, the results are unpredictable: the switchhook flash may be ignored or interpreted as the digit "1."

Authcodes after SSP can be stored as Speed Call or Autodial entries. When this is done, the stored number (entry) must contain only the access code and authcode digits.

All digits in the entry after the access code are interpreted as authcode digits. In the case of Authcode Conditionally Last, authcodes can be stored as Autodial entries, but not Speed Call entries. If necessary, the caller can continue to enter more authcode digits after operation of the Autodial or Speed Call key. However, for security reasons, authcodes should not be stored as Autodial or Speed Call entries.

Call Detail Recording If the Call Detail Recording (CDR) of authcodes is specified, then each time an authcode is entered, a record is generated on the CDR device. The record is passed to CDR only if one of the following occurs:

- the call becomes established, for example, trunk is seized or local telephone answers
- the call cannot be completed, for example, no trunks available

Authcode input by means of Tie trunks Authcodes can be entered by means of access tie trunks. Incoming or two-way tie trunk groups at a switch equipped with the NAUT feature can be defined to prompt for an authcode entry.

Direct Inward System Access If a caller makes a NARS call in association with a valid Direct Inward System Access (DISA) call, the NCOS associated with the DISA Terminal Number (TN) is used for NARS route selection. If the FRL of this NCOS is too low to access the route list that NARS has selected for the call, the caller will be prompted for an authcode entry, unless an authcode (for example, Authcode after SSP) was entered previously.

Barge-In/Busy Verify If Barge-In or Busy Verify is used by the attendant to break into a connection where an authcode is being entered, the authcode entry will be affected. If the authcode entered is invalid as a result, the user will be given overflow tone when the EOD timer expires.

Centralized Attendant Service The Central Attendant Service (CAS) feature enables several remote switches to share the attendant services at one central location. A CAS attendant can enter an authcode by means of a Release Link Trunk (RLT) before connecting or transferring calls to the connecting remote PBX.

If the CAS attendant enters a NARS number by means of a Release Link Trunk (RLT), the NCOS associated with the NCOS of the attendant at the remote PBX is used in the NARS route selection process. If the FRL of this NCOS is inadequate, the CAS attendant may be prompted for an authcode entry. See *Centralized Attendant Service description and engineering* (553-2681-100).

Call Forward The Call Forward feature provides two customer options:

- Call Forward-Originating party's CLS (CFO)
- Call Forward-Forwarding party's CLS (CFF)

With the NAUT package and the CFO option, a caller may be prompted for an authcode entry after a call to a telephone that forwards the call to a NARS number. With the CFF option, the user will not be prompted by the local switch for an authcode entry after such a call.

Network Class of Service An NCOS is assigned to each telephone and each incoming tie trunk at an ESN node. An authcode entry modifies the user's NCOS for the duration of the call. The FRL associated with the user's assigned NCOS is used to determine if it is necessary to prompt for an authcode entry. After an authcode is collected and validated, the NCOS associated with the authcode is used for the duration of the call.

Network Alternate Route Selection During NARS route selection, the FRL associated with the call originator's NCOS is compared with the FRL of the selected route list. If the originator's FRL is lower and no authcode was entered previously, the system may prompt for an authcode entry. A valid authcode modifies the originator's NCOS, and hence, FRL. This new FRL is then used for route selection.

Network Queuing When an authcode is entered, the NCOS associated with the authcode is used to determine Network Queuing capabilities.

Coordinated Dialing Plan Authcode after SSP can be used before dialing a CDP call. If the NAUT package is equipped, the "conditionally last" request for an authcode entry applies.

Requirements

Users on PBX or Centrex systems connected by tie trunks to an ESN node can use the Authcode Conditionally Last feature, provided that these systems transmit or repeat all digits dialed by the users in response to the authcode request. This feature cannot be used by certain systems that operate in senderized mode. Correct operation may require adjustment of EOD timeout on systems that employ simulated cut-through operation.

In an ESN network consisting of multiple switches equipped with the NAUT package, authcodes should be requested only once on a given call. This requires careful engineering of:

- the tie trunk group option for authcode prompting
- the FRL values assigned to route lists

Users at an ESN main or Conventional main arranged for the UDP by a dedicated trunk group to an ESN node can use the Authcode Conditionally Last feature at the ESN node in the same manner as those stations located directly at the ESN node. However, these users cannot access the Authcode after SSP feature by the same trunk group.

Coordinated Dialing Plan

The Coordinated Dialing Plan (CDP) allows a customer with a number of local switches to coordinate the dialing plan of the stations at these switches. A telephone user can call any other telephone within the CDP group of switches by dialing a unique three- to seven-digit number assigned to the telephone.

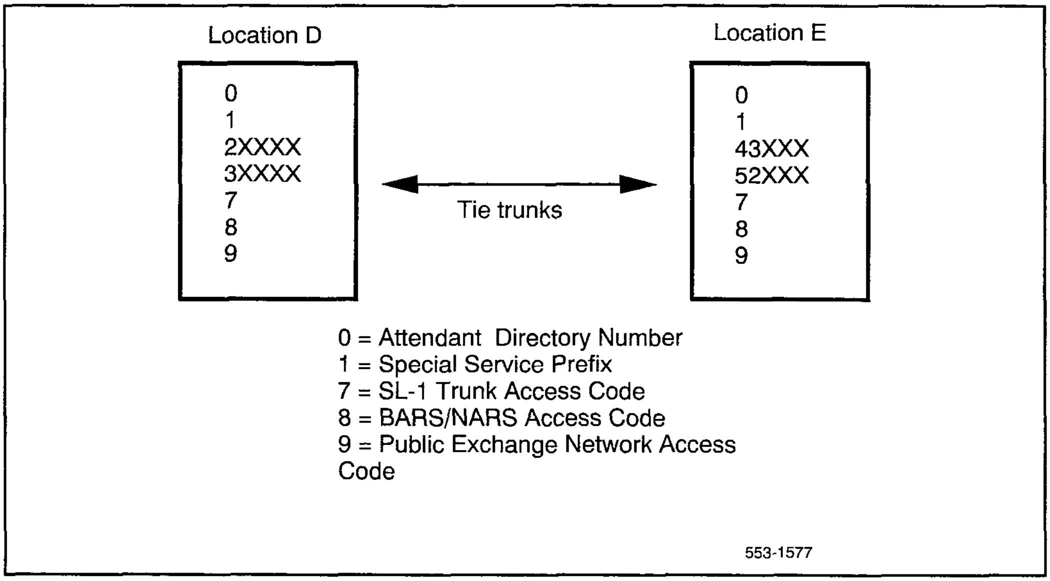
If the switch is equipped with the Directory Number Expansion (DNXP) package, the number assigned to each telephone can have up to ten digits.

A CDP can be arranged to provide a centralized public exchange network capability that channels access to and from the public network through a single switch within the CDP group.

The CDP software provides the translation and digit manipulation capability that is necessary to implement the coordinated dialing plan. Calls dialed within the CDP format can be terminated locally after digit translation and digit deletion. Or, calls can be routed to a remote switch in the CDP group following digit translation, route selection, and digit deletion and/or insertion. Figure 7 illustrates how a coordinated dialing plan might be implemented at two customer locations.

> *Note:* The maximum number of leading digits to be deleted from a Local Steering Code is four. However, if the Directory Number Expansion (DNXP) package is equipped, this number is increased to seven digits.

Figure 7
Example of a Coordinated Dialing Plan



Steering codes

Referring to Figure 7, users at Location D can call stations at Location E by dialing 43XXX or 52XXX. Similarly, users at Location E can call stations at Location D by dialing 2XXXX or 3XXXX. If a user at Location D dials 43XXX or 52XXX to reach a telephone at Location E, Location D uses the digits "43" or "52" as a Distant Steering Code (DSC) to select the trunk group to Location E. Similarly, if a user at Location E dials 2XXXX or 3XXXX to reach a telephone at Location D, Location E uses the digit "2" or "3" as a DSC.

The same format is used for calling local stations; for example, users at Location E dial 43XXX or 52XXX to reach local stations at Location E. In this case, the software interprets the digits "43" or "52" as a Local Steering Code (LSC) and deletes them from the dialed number in order to terminate the call locally.

Note: Where possible, four-digit extension numbers should be maintained as CDP DNs. As long as the first digit or digits of these extension numbers are unique at each location, all or part of the extension number may be used as a steering code. There cannot be duplicate extension numbers at two locations on a CDP network.

If the switch at Location E is arranged to provide centralized access to the public exchange network, digit 9 at Location E is considered a Trunk Steering Code (TSC) for public exchange access. At Location D, digit 9 is a TSC that invokes digit manipulation to insert the required digits to route the call through Location E to the public exchange network. Similarly, users at Location D can call the attendant at Location E by simply dialing "0", if Location D does not assign Digit 0 as the local attendant access code.

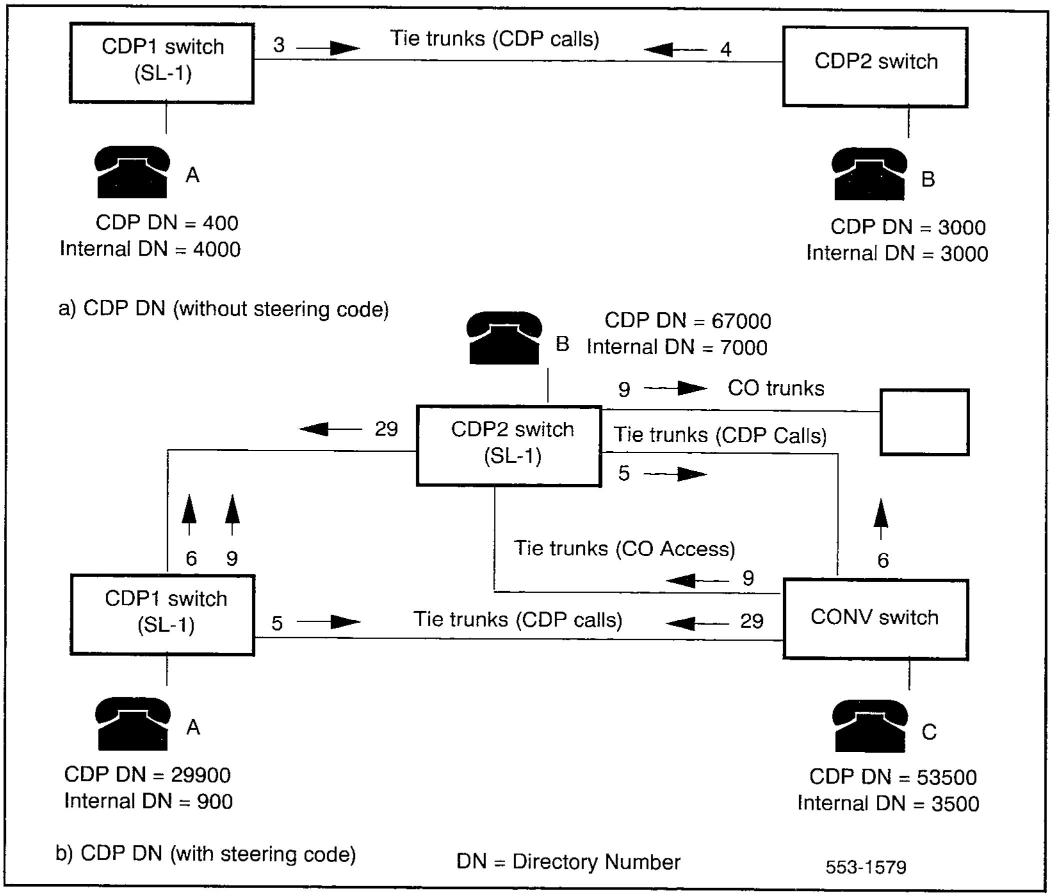
Steering codes can be composed of one to four digits. If the DNXP package is equipped, the steering codes can have up to seven digits.

At each switch in the CDP group, the steering codes must be distinct (for example, the initial digits must be unique) from any assigned access codes. As Figure 7 shows, "0" is reserved as the attendant access code; "1" is reserved as the Special Service Prefix (SPRE); "7" is reserved as a trunk access code; "8" is reserved as a NARS access code; and "9" is reserved as the public exchange network access code. This means there are five digits remaining that can be used as the leading digits of steering codes (for example, "2," "3," "4," "5," and "6"). The CDP feature supports up to 10,000 steering codes.

Note: Pre X11 release 13, the CDP feature supports up to 5000 steering codes.

A CDP DN consists of an internal DN prefixed with the appropriate steering code. A typical CDP configuration is shown in Figure 8.

Figure 8
A typical CDP configuration



Conventional switch access

If a Conventional (CONV) switch, any type without the CDP software, is integrated as part of a CDP group (see Figure 8), the steering codes defined at a CDP switch to access the Conventional switch may be repeated or absorbed (for example, deleted) by the CDP switch. The steering codes are repeated if the Conventional switch is identified by more than one steering code; they are absorbed if all the numbers at the Conventional switch begin with the same steering code.

Calls to a CDP switch from the Conventional (CONV) switch are made by dialing the desired CDP DN (for example, telephone C at the CONV switch dials 67000 to reach telephone B at switch CDP2). The CONV switch uses digit 6 as a trunk access code for the tie trunk route to switch CDP2. After tie trunk seizure, the CONV switch outpulses the remaining digits (7000) to CDP2. At CDP2, digit 6 is inserted on the incoming tie trunk from the CONV switch, and the call is completed to telephone B.

Local calls at the Conventional (CONV) switch are made by dialing only the internal DN (for example, 3500), rather than the CDP DN (for example, 53500), unless the CONV switch can be arranged to absorb digit 5, or employs a five-digit numbering plan.

As shown in Figure 8, switch CDP2 is arranged to provide centralized access to the public exchange network. For users at the CONV switch to access this capability, a separate tie trunk route must be provided to switch CDP2. This is because switch CDP2 is arranged to insert digit 6 on the incoming tie trunk route from the CONV switch used for CDP calls. For public exchange network calls, digit 9 must be inserted on the incoming tie trunk route from the CONV switch. Similarly, if users at the CONV switch are to be allowed access to the ESN capabilities (for example, NARS) at switch CDP2, another tie trunk route must be provided for this purpose.

A second alternative exists where one tie trunk route connects the CONV switch to the CDP equipped switch. Users dial an access code to this tie trunk group, then dial the necessary CDP digits to reach other switches, or the necessary BARS/NARS digits for network calls through the node. The tie trunk route at the CDP switch is not programmed to insert digits in this example.

CDP routing

Up to 128 route lists can be defined at a switch equipped with the CDP feature software. If CDP is equipped at an ESN node, 256 route lists can be defined. A route list is used to define the alternate route choices for CDP calls to a particular destination.

Route choices in a route list are called route list entries. There can be up to seven (0 to 6) route list entries associated with each route list. If a switch is equipped with the NARS software in addition to the CDP feature software, NARS route lists, a maximum of 256, can be shared by both NARS and CDP calls.

> *Note:* Pre X11 release 13, up to 32 route lists can be defined at a switch equipped with the CDP feature software. Also, there can be up to three (0 to 2) route list entries associated with each route list.

Route lists are associated with each Distance Steering Code and Trunk Steering Code that can be dialed at a CDP switch. Each code is defined to the CDP software, together with the route list number that must be accessed for call completion to the destination indicated by the steering code. The entries in the specified route list are then searched sequentially for an available and eligible trunk route. Local Steering Codes are not associated with route lists.

CDP digit manipulation

Route list entries can be associated with digit manipulation tables. There can be 32 (0 to 31) digit manipulation tables defined at a CDP switch. If CDP is equipped at an ESN node, 256 (0 to 255) tables can be defined. Digit manipulation table 0 is used as an indication that no digit manipulation is required.

Each digit manipulation table (except 0) can be defined to delete a number (0 to 15) of leading digits of a dialed CDP number and to insert up to 24 different leading digits, including the asterisk (*) to indicate a dialing pause, where required.

CDP Time of Day schedules

Two (0 to 1) TOD schedules can be defined at a CDP switch. If CDP is equipped at an ESN node, eight (0 to 7) TOD schedules can be defined. Each route list entry is associated with a TOD schedule. When a route list entry is selected for a CDP call, the CDP software compares the current time of day with the TOD schedule assigned to the route list entry.

If the current time of day is within the interval defined by the TOD schedule, the route list entry is used for the call. If the current time of day is not within the interval defined by the TOD schedule, or if the TOD schedule is turned "off" (in software), the route list entry is not eligible for the call. TOD schedules can be selectively turned "on" or turned "off" by the customer through service change.

Queuing

Queuing against local stations is provided by the standard Ring Again (RGA) feature. For calls directed to a remote CDP switch, RGA can be applied if all local outgoing trunk routes to the remote CDP switch are busy or blocked. RGA cannot be applied against busy or blocked stations/trunks at the remote CDP switch. Blocking tone is not provided until the full CDP number (or Trunk Steering Code [TSC]) is dialed.

Feature interactions

NARS The CDP feature can be implemented at a switch equipped with the NARS software feature. If such is the case, the following considerations apply:

- Steering codes for CDP calls must be distinct from the assigned NARS access codes.
- CDP numbers can be integrated with the ESN UDP; for example, a five-digit CDP number can be the same as the last five digits of a seven-digit UPD number.
- NARS/BARS route lists, digit manipulation tables, and TOD schedules can be shared by CDP calls. (CDP route lists must be numbered 0 to 31.)
- Users eligible for the OHQ and CBQ features can use them when placing CDP calls.
- FCAS does not apply to CDP calls.

Automatic Identification of Outward Dial (AIOD) and Automatic Number Identification (ANI) Calls made to the public exchange network when the AIOD or ANI feature is equipped will have either the internal DN recorded, if the call originates at the CDP switch interfacing to the public network, or the trunk access code, if the call originates at another CDP switch.

Attendant features If a user at a local CDP switch calls the local attendant, the local user's internal DN (not the full CDP DN) is displayed.

If a user at a CDP switch calls an attendant at another CDP switch, the trunk access code and member number of the incoming trunk are displayed. The following attendant features are supported at the local CDP switch, but are not supported between CDP switches:

- Automatic Timed Recall
- Barge-In, Busy Verify
- Camp-On
- Interposition Calling

CLS/TGAR treatment For CDP calls, all Class of Service (CLS) treatment remains the same as standard treatment with the exception of Conditionally Toll-Denied (CTD) and Conditionally Unrestricted (CUN) CLS, which are treated as Unrestricted (UNR). Users with an FR2 class of service can make local CDP calls but cannot make CDP calls to distant switches. Trunk Group Access Restrictions (TGAR) are ignored for routing CDP calls.

Code Restriction Code Restriction is applied to calls made only from stations with a Toll-Denied (TLD) class of service. Standard or flexible Code Restriction can be applied, on a trunk route basis, to public exchange network trunk calls.

Call Detail Recording The local internal DN (not the complete CDP DN) is recorded in the normal CDR manner. The maximum internal DN length remains at four digits; but if the DNXP package is equipped, the internal DN can have up to seven digits. The full CDP DN is shown in the dialed number field.

Common Control Switching Arrangement A CDP number can be part of a Common Control Switching Arrangement (CCSA) dialing plan. Digit absorption and manipulation for CCSA calls is handled as usual by the switch. A CCSA call can terminate at a switch in a CDP group other than the switch that hosts the CCSA network. This operation is transparent to the originator of the CCSA call.

Direct Inward Dialing Because a CDP DN can be up to seven digits long, the capability of inserting up to six leading digits on DID trunks is provided.

End-to-End Signaling End-to-End signaling is allowed for CDP calls.

Call Modification Call Modification, for example, Call Transfer (XFER), Call Forward, and Conference, is allowed for CDP calls. When using these features, the user dials within the CDP format.

Hunting Hunting across different switches in a CDP group is not supported. Standard Hunting can be applied to local CDP calls.

Message Center The Message Center capability is not supported across CDP switches. However, locally it operates as normal.

Digit Display

- **Outgoing CDP call** The complete dialed CDP DN is displayed at the originating telephone.
- **Incoming CDP call** The trunk access code and member number of the incoming trunk route is displayed.
- **Internal CDP call** At the originating telephone, the complete dialed CDP DN is displayed. If the call hunts or is picked up by another telephone, the internal DN of the answering telephone is displayed. At the terminating telephone, the internal DN of the originating telephone is displayed.

Network traffic measurements

The Network Traffic (NTRF) feature provides traffic measurement data related to network performance and network traffic. The NTRF feature can be equipped at ESN nodes and ESN mains. Using this data allows the network manager to assess the effectiveness of the network and to identify specific areas of network operation where improvements are needed.

The network traffic measurements accumulated at a switch equipped with the NARS (ESN node), BARS (ESN main), or CDP feature encompass the following areas of operation (in addition to regular traffic measurements):

— Network Class of Service (NCOS)

— Routing

— Off-Hook Queuing (OHQ)

— Call-Back Queuing (CBQ)

— Coordinated Call-Back Queuing (CCBQ)

— Call-Back Queuing to Conventional Mains (CBQCM)

— Incoming Trunk Groups

Routing traffic measurements

A route list is a list of outgoing alternate trunk routes to a specific location from a switch.

Trunk routes in a route list are termed route list entries. The number of route lists/entries that can be defined at a switch depends on the features equipped at that switch. Table 8 lists the parameters for the different features and feature combinations. The values shown in parentheses are for X11 release 13 and greater.

Legend for Table 8:

NCOS = Network Class of Service

FCAS = Free Calling Area Screening

SDR = Supplemental Digit Restriction

Table 8
Summary of networking feature parameters

Parameter	BARS	NARS	CDP	CDP with BARS	CDP with NARS
NCOS Groups	0–99 (0–7)	0–99 (0–15)	0–99 (0–3)	0–99 (0–7)	0–99 (0–15)
Facility Restriction Levels	0–7	0–7	0–7	0–7	0–7
Digit Manipulation Tables	1–255	1–255	1–31	1–255	1–255
Route Lists	0–127	0–255	0–31 (0–127)	0–127	0–255
Route List Entries	0–31 (0–7)	0–31 (0–7)	0–6 (0–2)	0–31 (0–7)	0–31 (0–7)
FCAS Tables	1–127	1–255	–	1–127	1–255
SDR Tables	0–255	0–511	–	0–255	0–511
Steering Codes	– –	– –	1–10,000 (1–5,000)	1–10,000 (1–5,000)	1–5000

Note 1: If the NARS and BARS features are equipped in the same switch but for different customers, the highest parameter values apply to that switch; for example, if one customer has NARS and another customer has BARS, the NARS parameters apply to the BARS customer.

Note 2: If the New Flexible Code Restriction (NFCR) feature is equipped, the number of available NCOS groups is 8, see *X11 features and services* (553-3001-305). With X11 release 13 and later, this number is expanded to 100.

Note 3: NSIG provides 16 NCOS groups.

Note 4: Parameters in parentheses are for releases prior to X11 release 13.

The TFN001 routing measurements provide data related to route list utilization. The measurements show how often a route list was accessed, which entries in the list were used, and whether the call was successful in completing a selection or connection. Routing traffic measurements are available at ESN node and ESN main switches.

The routing traffic measurements contain the following statistics for each defined route list:

- **Route list requests** This measurement identifies the total number of call attempts for which the called destination translations identified this route list to attempt call completion.
- **Route list requests served without delay** This measurement reflects the total number of network calls that were routed without encountering blocking or queuing.
- **Expensive route acceptances** This measurement identifies the number of calls that were allowed to complete over an expensive trunk route after the Expensive Route Warning Tone (ERWT) was given.
- **Route list requests standard blocking** This measurement identifies the number of call attempts that could not be served because a route or queuing process was not available to a user. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant.
- **Route list entry usage count** This measurement identifies the number of calls that were routed successfully over a particular route (entry) in a route list. A count is maintained for each route list entry.

OHQ measurements

Traffic measurements for Off-Hook Queuing (OHQ) are associated with each route list and identify the utilization of the OHQ feature. The OHQ measurements are included with the routing traffic measurements (TFN001), and contain the following statistics for each route list:

- **Quantity of calls placed in OHQ** This measurement identifies the number of calls that attempted to use a route in the route list. But because facilities were not immediately available, the call was permitted to remain off hook to wait for facilities.
- **Average time in OHQ** This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (expressed in units of 0.1 second) represents the average time in the queue. Calls that timed out in the queue before a route was selected are also included in the average.
- **Quantity of calls abandoned from OHQ** This measurement identifies the number of calls that were placed in the OHQ but were abandoned. For example, the caller went on hook before a route became available or the time limit was reached.

CBQ measurements

Traffic measurements for CBQ are associated with each route list and identify the utilization of the feature. The CBQ measurements are included with the routing traffic measurements (TFN001) and contain the following statistics (for each route list):

- **Quantity of CBQ calls** This measurement identifies the number of calls that were offered CBQ and accepted the offer.
- **Average time in CBQ** This measurement identifies the average duration (in units of 0.1 second) calls remained in the CBQ. Calls that were canceled and calls that were served are included in this measurement.
- **Quantity of CBQ offerings** This measurement is a count of the number of calls that were offered CBQ, regardless whether or not the offer was accepted.
- **Quantity of CBQ user cancellations** This measurement identifies the number of calls that were removed from the CBQ on the call originator's request. For example, cancellation of the Ring Again feature.
- **Routing traffic report output format** The routing traffic measurements are output for each route list as shown in Table 9.

Table 9
TFN001 routing format

System ID		TFN001							
Customer number									
RLST	xxx	route list requests	route list requests served without delay	expensive route acceptance	route list requests standard blocking	not defined	not defined		
	RT		route list entry use	route list entry use	route list entry use	route list entry use	route list entry use	route list entry use	route list entry use
			SL1TD calls	SL1TD calls	SL1TD calls	SL1TD calls	SL1TD calls	SL1TD calls	
	OHQ	OHQ calls	time in OHQ	abandoned calls					
	CBQ	CBQ calls	average time in CBQ	CBQ offerings		CBQ user cancel			
Example									
0434		TFN001							
000									
RLST	000	00345	00344	00012	00000	00000	00000		
	RT		00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
			00000	00000	00000	00000	00000	00000	00000
	OHQ	00000	00000	00000					
	CBQ	00000	00000	00000	00000				

Note 1: OHQ and/or CBQ information is printed only if the feature is equipped and activated.

Note 2: The two fields not defined always show zeroes (0).

NCOS measurements

The TFN002 NCOS measurements are shown in Table 10.

Traffic measurements are collected for each defined NCOS group to indicate the grade of service, in terms of blocking and queuing delay, being provided by the system. If a grade of service is determined by the communications manager to be inappropriate for users in a particular NCOS group, then the communications manager can either reassign the users to another NCOS group, redefine the characteristics of the existing NCOS group, or change the routing parameters. NCOS measurements are available at ESN node and ESN main switches.

The TFN002 NCOS measurements contain the following statistics for each defined NCOS group:

– **Quantity of calls attempted** This measurement identifies the total number of call attempts generated by users in an NCOS group.

– **Routing requests served without delay** This measurement identifies the number of call attempts that were routed without encountering blocking or queuing.

– **Expensive route acceptance** This is a count of the number of callers who accepted an expensive route to complete a call.

– **Network Call Standard Blocking** This measurement identifies the number of call attempts that could not be completed because a route or queuing process was not available to a user. The blocked call may have been routed to overflow tone, a recorded announcement, or the attendant.

– **Quantity of calls refusing expensive routes** This measurement identifies the number of calls that were given ERWT and elected not to use the expensive route.

– **Quantity of calls placed in OHQ** This measurement identifies the number of calls that were placed in the OHQ.

– **Average time in OHQ** This measurement identifies the average duration that calls remained in the OHQ until a route became available. The value (in units of 0.1 second) represents the average time that calls were in the queue. Calls that timed out in the queue before a route was selected are also included in the average.

- **Quantity of CBQ calls** This measurement identifies the number of calls that were offered CBQ and accepted the offer.
- **Average time in CBQ** This measurement identifies the average time that calls waited in the CBQ for a route to become available. It includes calls that requested a cancellation, calls that were served, and direct Ring Again against trunks. The average time is expressed in units of 0.1 second.

Table 10
TFN002 NCOS report

Format							
System ID	TFN002						
Customer number							
NCOS	network class of service group	calls attempted	routing requests served without delay	expensive route acceptances	network call standard blocking	not defined	calls refusing expensive routes
	OHQ	OHQ calls	average time in OHQ				
	CBQ	CBQ calls	average time in CBQ				
Example							
0423	TFN002						
000							
NCOS	000	00207	00197	00000	00001	00000	00000
	OHQ	00007	00237				
	CBQ	00000	00000				

Note 1: OHQ and/or CBQ information is printed only if feature is equipped and activated.

Note 2: The field that is not defined always shows all zeroes (0).

Incoming trunk group measurements

The incoming trunk group measurements (TFN003) are output as shown in Table 11.

The TFN003 Incoming Trunk Group Measurements provide an indication of the incremental traffic that was imposed on incoming trunk groups by the network queuing features. Data are provided for each incoming or two-way trunk group that is offered OHQ, CCBQ, or CBQCM. These measurements are available at ESN nodes.

The following measurements are accumulated for each incoming (or two-way) trunk group:

- **Quantity of calls placed in OHQ** This measurement identifies the number of incoming trunk calls that were placed in the OHQ for possible connection to another trunk group.
- **Average time in OHQ** This measurement reflects the average time (in units of 0.1 second) that calls waited in the OHQ for a trunk to become available. The average time includes those calls that were removed from the OHQ by caller abandonment or were removed from the queue after expiration of the OHQ time limit.
- **Quantity of incoming calls offered CCBQ or CBQCM** This measurement identifies the number of incoming trunk calls that were blocked at the ESN node and for which the user was given the option of accepting an ESN node–initiated call back when facilities would become available. The measurement relates to use of the CBQ feature by users at an ESN main (Coordinated Call-Back Queuing) or Conventional main (Call-Back Queuing to Conventional Mains).
- **Quantity of calls accepting CCBQ or CBQCM** This measurement identifies the number of incoming trunk calls that were blocked at the ESN node, were offered CBQ, and accepted the offer. The count relates to CBQ acceptances by users at an ESN main or Conventional main.
- **Average time in CBQ** This measurement (expressed in units of 0.1 second) reflects the average time that users at an ESN main or Conventional main remained in the CBQ (at the ESN node) for a facility to become available.

Note 1: When a CCBQ call back is offered to a busy station at an ESN main, the call is removed from the queue for 5 minutes, then reinserted in the queue. This process occurs only once. The additional queuing time is added to the average time. The 5-minute suspension time is not included in the average time, nor is its reinsertion into the queue pegged as another CBQ call.

Note 2: When a CBQCM call back is offered to a station at a Conventional main that is busy or fails to answer the call back, the call is removed from the queue and reinserted into the queue as specified in Note 1.

- **Quantity of calls blocked in call back** This measurement identifies the number of CBQ call backs (CCBQ or CBQCM) initiated by the ESN node that could not be completed because an outgoing trunk group (to the ESN main or Conventional main) was not available.
- **Callback Attempts No Answer and cancellation** This measurement identifies the number of call back attempts that were not successful because the caller failed to answer the call back. CBQ call backs to a station at an ESN main that has previously canceled CBQ are treated as Call-back Attempts No Answer.

Table 11
TFN003 Incoming Trunk Group

Format						
System ID		TFN003				
Customer number						
TRKG	incoming trunk group					
	OHQ	calls placed in OHQ	average time in OHQ			
	CBQ	incoming calls offered CBQ, CCBQ, CBQCM	calls accepting CBQ, CCBQ, CBQCM	average time in CBQ, CCBQ, CBQCM	blocked CBQ, CCBQ, CBQCM call backs	call back attempts not answered or canceled
Example						
0423	TFN003					
000						
TRKG	003					
	OHG	00006	00263			
	CBQ	00000	00000	00000	00000	00000

OHQ threshold violation measurement

The output format for this threshold measurement is shown in Table 12.

The OHQ overflow threshold measurement (TFN101) provides an indication that more than the expected number of users are timing out in the OHQ. This means that OHQ is offered and accepted, but a trunk does not become available before the service-changeable OHQ time limit expires. This could result from trunks being out of service, an incorrectly defined OHQ time limit, or temporary traffic overload.

Table 12
OHQ threshold violation measurement

Format	
System ID	TFN101
Customer number	
OHQT timed out OHQ calls	threshold
Example	
0423TFN101	
000	
OHQT 00333	00000

Traffic measurement options

New traffic measurement options are introduced with the NTRF feature.

These options are set and/or queried through use of the Traffic Control (TFC) program (LD02) in the normal manner. For more information, see *Traffic measurement formats and output* (553-2001-450). The options are:

- to generate routing measurements (TFN001)
- to generate NCOS measurements (TFN002)
- to generate incoming trunk group measurements (TFN003)

List of terms

AC

access codes

AIOD

Automatic Identification of Outward Dial

ANI

Automatic Number Identification

AUB

Authcode Data Block

AUT

Authcode Table

BARS

Basic Alternate Route Selection

BAUT

Basic Authorization Code

CAS

Central Attendant Service

CBQ

Call-Back Queuing

CBQ(i)

initial CBQ option

CBQ(a)

extended CBQ option

CCBQ

Coordinated Call-Back Queuing

CCBQAM

Coordinated Call-Back Queuing Against Main

CCSA

Common Control Switching Arrangement

CDP

Coordinated Dialing Plan

CDR

Call Detail Recording

CFF

Call Forwarding-forwarding party's COS

CFO

Call Forwarding-originating party's COS

CO

Central Office

CONV

Conventional (switch)

DDD

Direct Distance Dialing

DISA

Direct Inward System Access

DMI

Digit Manipulation Index

DN

Directory Number

DNXP

Directory Number Expansion

DSC

Distance Steering Code

DTMF

Dual Tone Multi-Frequency

EOD

End-of-Dialing

ERWT

Expensive Route Warning Tone

ESN

Electronic Switched Network

FCAS

Free Calling Area Screening

FCI

Free Calling Index

FRL

Facility Restriction Level

HLOC

Home Location Code

HNPA

Home Numbering Plan Area (code)

ITEI

Incoming Trunk Exclusion Index

ITGE

Incoming Trunk Group Exclusion

LDN

Listed Directory Number

LOC

Location Code

LPK

Loop key

LSC

Local Steering Code

NARS

Network Alternate Route Selection

NAUT

Network Authorization Code

NCOS

Network Class of Service

NCTL

Network Control

NPA

Numbering Plan Area (code)

NSC

Network Speed Call

NSIG

Network Signaling

NTRF

Network Traffic (measurement)

NXFER

Network Call Transfer

NXX

Local Exchange

OHQ

Off-Hook Queuing

RDB

Route Data Block

RGA

Ring Again

RLB

Route List Block

RLT

Release Link Trunk

RTC

Routing Control

SAT

Satellite Link Control

SDRR

Supplemental Digit Restriction/Recognition

SPN

Special Number

SPRE

Special Service Prefix

SSC

System Speed Call

SSP

Special Service Prefix

STD

Standard (signaling)

TCM

Traveling Class Mark

TCOS

Traveling Class of Service

TFC

Traffic Control

TFN

Network Traffic

TGAR

Trunk Group Access Restrictions

TLD

Toll Denied

TOD

Time of Day

TSC

Trunk Steering Code

TTTN

Tandem TIE Trunk Network

UDP

Uniform Dialing Plan

XFER

Call Transfer

SL-1

Electronic Switched Network

Description

Release 6.0
Standard
October 31, 1993
Printed in the U.S.A.

northern telecom

SL-1

Electronic Switched Network

Signaling guidelines

Publication number: 309-3001-180
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Introduction

The Electronic Switched Network (ESN) is a private communications network intended for use by large business customers with distributed operation locations. See *ESN description* (309-3001-100).

This Northern Telecom Publication describes the signaling guidelines and considerations that are applicable to ESN networks. Information presented here can be used to determine the compatibility of a switch that is to be incorporated as part of an ESN network. A companion publication, *ESN transmission guidelines* (309-3001-181), should be used in conjunction with this publication to determine transmission requirements.

PBX networks

PBX networks have three basic elements:

- telephones
- PBX switch
- transmission facilities

Each telephone is connected to a PBX switch which establishes connections between the telephones. Connections between telephones on different switches in a PBX network are established over a transmission facility (i.e., trunk) between the pair of switches, or over a tandem connection of transmission facilities and intermediate switches.

A user establishes a call to another telephone by dialing a string of digits which direct the connection to that telephone. In Tandem Tie Trunk Networks (TTTN), the dialed digits steer the connection through the network. A string of successive access codes cause facilities to connect in tandem until the switch which has the destination telephone is reached. Each time a switch connects a new facility in tandem, it passes on the digits to the connected facility.

In a Meridian 1 network, the ESN node switches collect all digits of the called number, and pass the full called number between switches. The called number is dialed in a Uniform Dialing Plan (UDP) format. Users at switches connected to Meridian 1 ESN nodes may have to dial in a TTTN format to reach the node before dialing the called number in UDP format. Similarly, the ESN node may control routing to connected switches by transmitting a sequence of access codes, followed by the called number.

Switches are also provided with trunks to the public network. Calls to telephones in the public network (and to other switches) can be placed by these trunks. However, trunks between switches (tie trunks) are normally provided because the calling can be performed at a lower cost.

PBX designations

PBX switches in a network configuration have designations that depend on the network function. Designations used include node, main, satellite, and CCSA tandem.

Node switch

Node is the designation in an ESN network for a Meridian 1 switch with the ESN Network Alternate Route Selection (NARS) software package. The Meridian 1 ESN node has the full capabilities for selecting trunk routes, while other non-mode switches have only limited route selection capabilities.

Main switch

A Main Switch is a switch which has an outgoing trunk route to only one node. This route may be both incoming and outgoing, but routes to other nodes must be incoming only to the Main PBX, since the Main PBX cannot select between nodes for the routing of network calls. A Main PBX may have CO, FX, and WATS trunks, as well as tie trunks to other switches.

A PBX which has outgoing trunk routes to more than one node, but does not meet the requirements to be classed as a node, is also classed as a main. The outgoing routes may be assigned different access codes, leaving route selection to the user, or may be selected by an automatic route selection capability.

Tributary switch

A tributary switch is a PBX which has a trunk route to a Main PBX, but not to a node. A tributary PBX may have CO, FX, and WATS trunks, as well as tie trunks to other switches.

Satellite switch

The only difference between a satellite and tributary PBX switch is that the satellite switch has neither incoming Central Office (CO) trunks nor attendant service. In this document, satellite and tributary PBX switches are treated the same.

CCSA tandem switch

A Common Control Switch Arrangement (CCSA) tandem switch is the major switching vehicle in a CCSA network. The tandem switch performs trunk-to-trunk switching only. Its function is similar to that of a Meridian 1 ESN node, the difference is that a Meridian 1 ESN node can support telephones in addition to trunk-to-trunk switching. Switches with tie trunks to the tandem switch are called Main switches. The CCSA network supports a dialing plan similar to the ESN Uniform Dialing Plan.

The CCSA tandem switch is the interface point between the ESN network and a CCSA network. The two networks can be arranged to function as a single integrated network.

Supervision

Supervision is a binary signal associated with each direction of transmission on a trunk facility. The two states are on-hook and off-hook, analogous to condition of a telephone on-hook (hung up) or off-hook (in use).

Each switch connected to a trunk sends a supervision signal to, and receives a supervision signal from the connected switch. Thus, the trunk has four supervision states. The trunk is idle when both directions are on-hook. Off-hook is sent when a call is initiated on an idle trunk. This action is called "seizing" the trunk. The distant switch receives the off-hook signal, and prepares to receive digits. A momentary off-hook condition returned from the destination switch may occur during call set up, but a steady off-hook is not transmitted until the called telephone answers. This off-hook signal is called "answer supervision". The supervision changes to on-hook when the called telephone hangs up or is disconnected.

When a switch serves as an intermediate (tandem) connection between trunks, it normally sends the supervision signal it receives from each incoming trunk to the connected outgoing trunk. It also monitors for a disconnect signal so the trunks can be returned to an idle state and be ready for new calls.

In some cases, there are not any provision made to return an answer supervision signal from the destination when the called telephone answers. Under this condition, an off-hook signal can be returned from an intermediate switch. This off-hook signal is called "substitute answer supervision". This signal is provided to remove transmission impairments associated with the on-hook condition on some trunk facilities and to distinguish a call which has been blocked from one which might have reached a destination party.

A connection by a public network trunk. is a typical case where a called telephone answer indication is not returned, is Even if provisions can be made to provide an indication of answer, it normally isn't. The PBX switch which connects the call to the public network trunk should be configured to provide an off-hook signal to the incoming trunk after sending digits to the public network trunk.

Digit transmission

In an ESN network, digits of the called number are transmitted between pairs of switches in one of two modes: Dial Pulse (DP) or Dual Tone Multifrequency (DTMF).

In Dial pulse, each digit is represented by a string of pulses. The digit zero is represented by ten pulses. Each other digit is represented by the corresponding number of pulses. The pulses are transmitted as interruptions of direct current.

In DTMF, digits are transmitted over the speech path by a tone code. Digit transmission takes place at a higher rate than dial pulse (typically two to ten times faster).

Modern PBX switches are compatible with either mode. Older equipment is only compatible with dial pulse. ESN uses DTMF wherever practical, to take advantage of its higher transmission speed. However, dial pulse is sometimes required. Furthermore, there are some transmission impairments associated with DTMF. These impairments are normally only important when equipment is still connected after conversation begins to take place. The impairments are removed when the receiving equipment is disconnected.

Modes of operation

A PBX switch can operate in one of two modes when routing a call over a tie trunk.

- cut-through mode
- senderized mode.

Cut-through operation

With cut-through operation, a trunk is accessed immediately following an access code. Subsequent digits are forwarded to the trunk as dialed. The telephone user monitors call progress tones from connected switches. The user may be required to pause in dialing to monitor for dial tone, or may be required to abandon prior to completing dialing due to blocking tone.

Pure cut-through operation provides the greatest flexibility for providing compatible operation for calls originated at main and tributary switches. The number of digits transmitted to the node can be flexible. The node can prompt for additional digits when required for the authorization code features.

DTMF to DP conversion

A variation on cut-through operation is to provide DTMF to dial pulse conversion. The user dials DTMF digits, but dial pulses are transmitted to the trunk. The converter may block transmission in the caller to called party direction while waiting for digits. The converter may detach to remove these transmission requirements. Since the last digit can not be easily detected, timing is often used to determine the end of dialing.

When the switch has completed outpulsing, it waits a specifies time for additional digits. If there are not any digits received in that interval, conversion is disabled. The timing may not properly distinguish a pause in dialing from a last digit, prematurely cancelling the forwarding of digits. Extending the timing causes the transmission impairments to exist during conversation if the called party answers quickly.

The Meridian 1 ESN nodes combine cut-through and senderized modes of operation. The switch collects the access code and enough digits to select a trunk. During this interval, the operation is very close to a register sender mode. The trunk is accessed and a string of digits outpulsed (not necessarily the same as those dialed). Subsequent digits are forwarded to the trunk as dialed, in a receive and resend mode, which is closer to cut-through operation.

Senderized operation

With senderized operation, all digits of the called number are collected before an outgoing trunk is accessed. The trunk is accessed, and digits are transmitted to set up the call. The transmitted digits need not be the same as those dialed (i.e., as a result of NARS digit manipulation). The user does not receive call progress tones until all digits have been transmitted. No tones are provided during dialing other than a locally generated dial tone following the trunk access code.

Senderized operation limits flexibility. The main PBX must be programmed to determine how may digits to collect before forwarding those digits to the node. Usually, no more than 12 digits may be collected and forwarded.

Outpulsing control

After a tie trunk is seized, digits of the called number will normally be transmitted. The only exception is a manual trunk, which rings a designated telephone when seized. However, most terminating equipment requires a variable time interval to prepare for reception of digits. This time interval often depends on the switch's call-processing load. Therefore, a fixed delay before sending digits would be unreliable. There are four commonly used ways of handling start dial control.

Immediate start

Applies in those cases where a short fixed delay is required for the switch to prepare to receive digits.

Delay for dial tone

A dial tone is provided when the switch is ready to receive digits.

Wink start

A momentary off-hook signal is sent when the terminating equipment is ready to receive digits.

Delay dial

An off-hook signal is sent to signify that the switch is not ready to receive digits. An on-hook signal is sent when the switch is ready to receive digits.

The delay for dial tone is used when a user controls digit sending, as in a TTTN. Wink start and delay dial are used in registerized digit sending. Normally, only one start dial control is used. However, on some switches, a.dial tone may be combined with any of the other three

The Meridian 1 switches in an ESN network are able to work with any of the start dial signals.

Call setup sequences in ESN

Following is a description of the various call setup sequences that can occu between conventional PBX equipment and a Meridian 1 ESN node, with emphasis on signaling compatibility. Calls are described (1) to the node, (2 between nodes and (3) from the node. Several cases are included, reflectin variations such as the type of machine involved (cut-through or senderized and whether or not tandeming is involved.

The Meridian 1 PBX without ESN main or ESN node software is treated a: a cut-through PBX in the following discussion. The Meridian 1 PBX with ESN main software is not addressed.

Calls to the node

Case 1

This case pertains to a call to the Meridian 1 ESN node from a telephone at a cut-through main switch.

- The user at the main PBX goes off-hook, receives dial tone from the main PBX, then dials the network access code.
- The main PBX seizes a tie trunk to the Meridian 1 ESN node and provides audio transmission to the caller. This permits subsequent call progress tones from the node to be heard by the caller.
- The node returns dial tone to the caller when it is ready to receive digits.
- The user dials the desired number and, if required, the authorization code. The digits are transmitted to the node as dialed. The node provides an Authorization Code Request Tone only if the authorizatior code is required. If not, the routing takes place immediately following the last digit of the called number.
- The node proceeds to set up the call.

Case 2

This describes a call from a cut-through non-senderized tributary PBX to the Meridian 1 ESN node by a cut-through non-senderized main PBX. Whenever practical, a direct trunk group to the Meridian 1 ESN node should be provided to avoid using the main PBX as a tandem switch.

- The user at the main tributary PBX goes off-hook, receives dial tone form the tributary PBX, and dials a tie trunk access code to access the main PBX.
- The tributary PBX seizes a tie trunk to the main PBX, and provides audio transmission so that tones from the main PBX can be heard by the user, and dialed digits from the user can pass to the main PBX. The main PBX normally provides dial tone to incoming tie trunks.
- After receiving dial tone from the main PBX, the user then dials the ESN network access code and desired number. The call setup proceeds as in Case 1 except that the user's dialed digits and the tones from the node pass through both cut-though switches.

Case 3

This describes calls from stations at a senderized main PBX to a Meridian 1 ESN node.

- The user goes off-hook, receives dial tone, dials the ESN access code, receives second dial tone from the main PBX, and then dials the desired number.
- The senderized main PBX does not seize a tie trunk to the node after receiving the access code. Instead it collects the digits of the called number, and then seizes a tie trunk to the node. The node does not provide dial tone.
- The senderized main PBX and the node are mutually arranged to utilize either a wink-start or delay-dial signal as a start-dial signal to initiate outpulsing.
- When the main PBX receives the appropriate start-dial signal, it outpulses the called number to the node, and then connects the user so that subsequent ringing signals can be heard.

The Meridian 1 authorization code is not supported in this case. The limitation which prohibit the authorization code are:

- The senderized main PBX has no provision to provide the authcode request tone for authorization code digits.
- The senderized main PBX does not have the capability to register enough digits for the authorization code and called number.
- Forwarding of additional digits after the senderized main PBX has outpulsed the called number to the node is not practical.

Case 4

This applies to calls from a cut-through tributary PBX to the Meridian 1 ESN node by a senderized main PBX. This case is similar to calls from a senderized main PBX. The user at a cut-through tributary PBX dials a tie trunk access code to reach the main PBX and receives dial tone from the main PBX. From this point on, the call is handled as though it originated at the senderized main PBX.

Case 5

This applies to call from a senderized tributary PBX to the Meridian 1 ESN node by a cut-through main PBX. This is not permitted because of signaling compatibility problems. The tributary cannot provide the proper outpulsing control for routing the call through the main to the node. Direct trunks must be provided from the tributary PBX to the node in this situation. By definition, the tributary then becomes a main PBX.

Case 6

In this case, both the main PBX and tributary PBX are senderized. This is also not permitted because of compatibility problems similar to Case 5. Direct trunks must be provided to the node from the tributary PBX. The tributary PBX thus becomes a main PBX.

Call completions

Call completions to stations on the node, stations at other nodes, and public network trunks form the node and from the connected nodes are handled in the normal manner.

The node completes calls, by tie trunks to main, to the following destinations:

- stations at the main PBX
- stations at a tributary PBX connected to the main PBX
- off-network stations by public network trunks terminating on the main
- off-network stations by public network trunks terminating on the tributary

In all cases, call routing to the main is initiated by an off-hook signal sent to the tie trunk. The basic sequences for call completion are:

- to reach telephone at main PBX, outpulse the Directory Number (DN)
- to reach a telephone at the tributary PBX, outpulse an access code for a tie trunk to the tributary, pause if necessary, and outpulse the DN
- To reach a public network telephone by a CO trunk terminating on the main, outpulse the access code for the CO trunk, followed by the public network number.
- To reach a public network telephone by the tributary PBX, the node outpulses an access code for the main to tributary tie trunk, followed by the access code to the CO trunk, followed by the public network number.

Case 1

This case applies to a call from a Meridian 1 ESN node to a telephone at either a cut-through or a senderized main PBX.

- The node seizes a tie trunk to the main and then pauses, waiting for the main to become ready to receive digits.
- The Meridian 1 then outpulses the telephone DN digits to the main PBX.

Case 2

This case involves a call from a Meridian 1 ESN node to a tributary PBX by a cut-through main PBX. This is similar to Case 1 except the initial digit(s) outpulsed by the node is an access code for a tie trunk connecting the main PBX to the tributary PBX. The procedure then is as follows:

- The node inserts a fixed pause (for delay for dial tone) after the access code, unless both main and tributary are step-by-step (SXS) switches.
- The node resumes outpulsing either when the fixed pause interval elapses or dial tone is detected.
- The resumed outpulsing is the DN at the tributary PBX.

Case 3

This case involves a call from a Meridian 1 ESN node to an off-net telephone by a cut-through main PBX.

- The access code outpulsed by the Meridian 1 ESN node is for a CO trunk, instead of tie trunk as in Case 2.
- A fixed pause or a delay for dial tone after the access code is required, even when both the main PBX and the CO are SXS switches.
- The resumed outpulsing consists of the public network number rather than a PBX number.

Case 4

This case applies when the main PBX is senderized and tandems a call from the Meridian 1 node to a tributary PBX or to a CO.

The beginning of the call setup sequence when the main PBX is senderized is the same as the previous case. However, once the node begins outpulsing, it outpulses all of the digits without pausing. If the main PBX can receive DTMF digits, outpulsing should be DTMF digits regardless of the capability of the tributary PBX or CO. The senderized main PBX collects the digits, translates, prefixes and completes the call to the next switch.

Calls to other private networks

Tandem tie trunk networks (TTTN)

Calls to a TTTN are engineered similarly to calls to a main PBX with tributaries. The maximum number of switches connected in tandem is five in a TTTN setup. Thus, up to three access codes with pauses.may have to be outpulsed. To avoid the requirement for a large number of digits to be outpulsed, trunks to several switches in the TTTN should be arranged so that no more than two trunks in tandem are required to reach any telephone in the TTTN.

Common control switching arrangement (CCSA)

A Meridian 1 ESN node with tie trunks to a CCSA switch is arranged to outpulse 7 digits to the CCSA switch to complete network calls to telephone in that part of the network. Optionally, the Meridian 1 ESN node can outpulse 10 digits to complete off-networked calls by the CCSA switch

Calls from other private networks

A Meridian 1 ENS node is able to function as a TTTN switch for calls from a TTTN. The user in a TTTN sequentially dials access codes to add trunks in tandem.

Calls from a TTTN do not require different engineering than calls from a main PBX, other than a Meridian 1 route data block option to arrange the incoming trunk for TTTN operation. TTTN operation is currently supported by Meridian 1 and is not changed for ESN.

Calls from a CCSA switch can be arranged to terminate on a Meridian 1 ESN node or any other switch which is part of the Coordinated Dialing Plan (CDP). Current Meridian 1 CCSA trunk options can accommodate this. Tie trunks from a CCSA switch must be provided to all switches in the ESN network (including those that are part of the CDP) to be accessible from the CCSA switch. An unambiguous numbering plan encompassing both the ESN network and CSSA must be arranged. The Meridian 1 ESN node routes calls in the CCSA numbering plan to the CCSA switch.

Requirements for dialing pauses

When outpulsing to main PBX and to a TTTN, the Meridian 1 ESN nodes are occasionally required to pause at various points in the digit strings to allow for trunk access and register attachment. Failure to pause causes digits to be missed, calls to be connected to wrong numbers or lost altogether.

The Meridian 1 ESN software provides for pauses following trunk access codes in the NARS translation tables. The general rule is that each trunk access code outpulsed must be followed by a pause. However, there are a number of situation where the pause is not required.

In determining whether the pause is required, you must consider:

- What type of PBX has been reached in the dialing.
- What piece of equipment is being accessed.

The situations where pauses are not required are:

- The access code is for connecting a step-by-step PBX to a step-by-step PBX.
- The access code is for connecting a Meridian 1 PBX to any other PBX, providing that subsequent pauses are not required.
- The access code is 9 for a CO trunk by a Centrex PBX, but no to other CO trunks.
- The access code is for an automatic route selection on any PBX.

A potential problem area is where a trunk access code requiring a pause is made after the call has been routed through one or more Meridian 1 switches. While the Meridian 1 ESN node need not pause between access codes for routing through the Meridian 1 switches, if it does not, a problem occurs at the point where the pause is required.

The Meridian 1 switches that are connected insert the proper delay after the access code before resending digits. However, the time spacing between digits is not maintained. The trailing digits "catch up" with the leading digits. The time delay required after an access code is eliminated. To avoid this problem, the Meridian 1 ESN node should insert pauses after each access code for this call routing.

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Contents

Introduction

The Electronic Switched Network (ESN) is a private communications network intended for use by large business customers with distributed operating locations. See *ESN general description* (309-3001-100).

This document describes the major transmission considerations that must be taken into account when planning an ESN network. The ESN transmission objective is to provide transmission quality comparable to public network calling. While this objective can easily be met for on-network calls, it is unreasonable to meet this objective for tandem combinations of public and private network trunks, since this produces a connection with inherently poorer performance. However, the sacrifice in performance should be small enough to be tolerable or unnoticeable. Noticeably poorer performance leads to an excessive number of calls placed by manually initiated bypass routing through the Direct Distance Dial (DDD) network.

Transmission considerations

Echo

All voice connections between telephones require two directions of transmission for conversation to take place. When the signal transmitted in one direction is reflected over the other directional path, the caller hears his/her own voice with a slight delay.

Depending on the delay, the effect is perceived as sidetone, rain barrel effect or echo. Two-wire facilities require care in matching impedances in order to prevent reflections. Four wire facilities do not generate reflections themselves, but they do not eliminate the reflection problem in built-up connections, since there are, in most cases, 2-wire connections to the telephones.

The objection to echo increases with the echo delay. The Via Net Loss (VNL) plan provides an increasing loss depending on delay. However, the loss also reduces the received volume. Limits are placed on the amount of loss used to suppress echo. When these limits are exceeded, echo suppressor devices can be used instead.

Loss

The provision of good transmission requires the following compromises.

- the need for sufficiently low (one-way) loss in each direction to provide satisfactorily high received volumes
- minimum contrast in received volumes on different calls
- the need for sufficiently high round-trip losses to ensure adequate performance from the standpoint of suppressing talker echo, noise, and near-singing

The following loss plan has been developed for ESN. The network is partitioned into node-to-node connections, node-to-main connections and main-to-satellite or tributary connections. The plan requires that:

- node-to-node trunks have a maximum loss of 3.5 dB
- node-to-node tandem connections have a maximum loss of 4.1 dB
- node-to-main trunks have a maximum loss of 2.5 dB

These loss objectives are met by installing echo suppressors and reducing the loss to 0 dB on trunks when the objective loss is exceeded with VNL alone.

Tandem switching

Public network tandem switching

A public network tandem switch is usually collocated with the carrier facilities that serve the switch. Tandem connections can be made between trunks terminating at this switch without significant degradation of transmission performance.

PBX network tandem switching

A PBX network tandem switch is usually located on customer premises, remote from the carrier facilities that serve the PBX switch. These carrier facilities are generally located at a telco switching center and connected to the PBX switch by local loop plant (cable). If the telco switching center does not have long-haul carrier facilities, short-haul carrier facilities are used to connect to another telco switching center which has these facilities. Thus a tandem connection made at the PBX switch, can introduce the distortion of two loop plant connections and two carrier facility connections, both avoided in the public network tandem connections.

Trunk routing rules

Trunk routing rules define the allowed connections between node, main, tributary, and satellite PBX switches. These routing rules are summarized in Table 1.

Table 1
Trunk routing rules

To From	Node	Main	Tributary	Satellite
Node	Yes (up to 4 links)	Yes (Note 4)	No (Note 3)	No (Note 3)
Main	Yes (one node only)	No (Note 1)	Yes (one main only)	Yes (one main only)
Tributary	No (Note 3)	Yes (one main only)	No (Note 2)	No (Note 2)
Satellite	No (Note 3)	Yes (one main only)	No (Note 2)	No (Note 2)

Note 1: Permitted for non-tandem trunks if the connected main PBX switches are part of a coordinated dialing plan. Also, routes of this type already in place when ESN is installed may be allowed to remain. However, these routes should be eliminated as part of network evolution, to support the dialing plan.

Note 2: Routes of this type already in place when ESN is installed may be allowed to remain. These routes should be eliminated as the network evolves.

Note 3: This route is permitted by upgrading the tributary or satellite switch to main switch capabilities.

Note 4: One-way routes from nodes to main PBX switches are not restricted.

Transmission planning

The planning of Electronic Switched Networks can be partitioned into four major tasks:

- assess the current equipment to identify locations of nodes, and designate existing equipment to remain in place as main, satellite and tributary PBX
- plan the tie trunk routes
- plan the network routing
- plan off-network call routing

Placement of nodes

The first decision is the choice of locations for node switches. The following transmission factors should be considered:

- The node should be located near a telco toll switching center to minimize the loop plant and short-haul facilities between the node and long-haul facilities.
- The node should be near the middle of its cluster of main PBX switches, so most main-to-node facilities fall in the short-haul category.
- If earth satellite facilities are used extensively, nodes should be located near the earth station facilities.

Once the nodes are designated, the remaining PBX switches are designated as main, satellite or tributary switches. All switches which have direct trunk groups to a node are main PBX switches. Those switches which access a node via main PBX are satellite or tributary switches, depending on whether or not they have incoming C.C. trunks.

Tie trunk routes

Each pair of nodes in the network represents a potential tie trunk route. Traffic considerations and tariffs determine how many routes are equipped and how many trunks are required for each route. In general, the objectives of least cost and best transmission both dictate that the number of tandem trunks required to establish any connection be kept to a minimum. Thus, direct trunk routes should be established wherever practical.

Direct trunk routes represent a radical departure from Tandem Tie Trunk Networks (TTTN). For TTTN, routing is usually organized into major trunk route "highways" with "feeder" routes. This structure is efficient when alternate routing is not permitted, as for TTTN, but it is not efficient when alternate routing, as for ESN, is supported. Thus, converting a large TTTN to an ESN can have large impact on permitted routing.

Network call routing

The direct and alternate routes used for routing calls must be planned so that transmission can be evaluated on each route and appropriate controls established on individual trunks to meet objectives. Table 2 gives the maximum number of node-to-node routes which may occur in networks of various sizes. The table establishes the upper limits on the number of routes to ensure that no potential route is overlooked.

Table 2
Route count table (maximum number of node-to-node routes)

Nodes	Direct routes	2 trunks in tandem	3 trunks in tandem	4 trunks in tandem
1	0	0	0	0
2	1	0	0	0
3	3	3	0	0
4	6	12	12	0
5	10	30	60	60
6	15	60	180	360
7	21	105	420	1260
8	28	168	840	3360
9	36	252	1512	7560
10	45	360	2520	15120
11	55	495	3960	27720
12	66	660	5940	47520

Assumptions:

— Each node has trunks to every other node.

— All routes that do not include the same node more than once are valid. (In practice, the number of valid routes is considerably less, since many of the routes will not make sense in a real work environment.)

Transmission controls

Transmission controls are first established for each direct trunk route; then two-trunk tandem routes; then three-trunk tandem routes; and so on. For each route, an Echo Suppressor (ES) control or VNL value is specified. (VNL applies to land circuits less than 1800 miles [2800 km]; longer land circuits and all satellite circuits require echo suppressor control.)

The required VNL value is determined from the round-trip delay which depends primarily on the type of facility and distance. Table 3 gives approximate VNL values for varying distances (airline miles) based on the type of facilities typically provided by a telco. The actual distance could be considerably greater since the actual signal path is less direct. However, margin has been added to accommodate average deviation from direct routing.

Table 3
Loss table for land circuits

Distance miles (km)	VNL (dB)
0–100 (0–160)	
100–400 (160–640)	1.0
400–700 (640–1120)	1.5
700–1000 (1120–1600)	2.0
1000–1300 (1600–2080)	2.5
1300–1600 (2080–2560)	3.0
1600–1800 (2560–2880)	3.5

Note: Land circuits longer than 1800 miles (2880 km) and all satellite circuits require echo suppressor control.

Two-trunk tandem routes

Wherever two VNL trunks are in tandem, their VNL losses are summed. If the loss exceeds 4.1 dB, at least one of the trunks should be equipped with echo suppressors if the route is to be permitted. Normally, the higher loss trunk is selected for echo suppression but another consideration is the minimization of the number of trunks which must be changed to echo suppressor control. Thus, trunks which appear most frequently in high-loss connections should also be considered prime candidates. Table 4 summarizes this requirement.

Table 4
Transmission control requirements for two-trunk tandem connections

Trunk 1	Trunk 2	Requirement
E.S.	E.S.	No action
E.S.	VNL	No action
VNL	E.S.	No action
VNL	VNL	If total loss is less than 4.1 dB, no action; otherwise, change trunk 1 to E.S. or prohibit this connection.

Three-trunk tandem routes

For smaller networks, all three-trunk tandem routes should be sorted and tested for proper controls. For larger networks, the number of three-trunk tandem routes actually permitted is small in comparison to the number of possible routes. Therefore, the more efficient approach is to sort out the permitted routes and to consider only them.

For three trunk tandem routes, echo suppressor combinations must be considered as well as total loss. Whenever a VNL tie trunk is connected with echo suppressors between two trunks , that trunk must be equipped with echo suppressors to make sure that intermediate echo suppressors are disabled.

Table 5 summarizes the transmission control requirements for three-trunk tandem connections.

Table 5
Transmission control requirements for three-trunk tandem connections

Trunk 1	Trunk 2	Trunk 3	Total loss If less than 4.1 dB	Total loss If greater than 4.1 dB
E.S.	E.S.	E.S.	No action required	No action required
VNL	E.S.	E.S.	No action required	No action required
E.S.	VNL	E.S.	Prohibit this connection (preferred) or change trunk 2 to E.S.	
E.S.	E.S.	VNL	No action required	No action required
VNL	E.S.	VNL	No action required	Change trunk 1 or 3 to E.S. or prohibit this connection
E.S.	VNL	VNL	No action required	Trunk 2 must be E.S. or prohibit this connection
VNL	VNL	E.S.	No action required	Trunk 2 must be E.S. or prohibit this connection
VNL	VNL	VNL	No action required	One or more must be E.S. or prohibit this connection

Note: When a trunk is changed to echo suppressor (E.S.) control all combinations in which it appears must be checked for routing violations.

Off-network call routing

The transmission properties of a call depend to a large extent on the type of facilities (Table 7) over which the call is transmitted. Because of these properties, certain destinations are disallowed for some types of off-network call routing. These are summarized in Table 6.

Table 6
Permitted off-network call routing

Route off-network call			Destinations permitted				
From	**By**	**By**	**Station**	**C.O. trunk**	**FEX trunk**	**WATS trunk**	**Other common carrier**
Node			Yes	Yes	Yes	Yes	Yes
Tie trunk	Node		Yes	Yes	Yes	Yes	Yes
Node	Main (note)		Yes	Yes	No	No	No
Node	Main	Tributary or satellite	Yes	No	No	No	No
Main	Tributary or satellite		Yes	Yes	Yes	No	No

Note: Off-network call to C.O. or FEX trunks are allowed only within local calling areas.

Network facilities

Private network facilities

Private network facilities can be ordered from the local telephone company and other common carriers. Although facilities are ordered on a point-to-point basis, the telco must be informed of the overall planned network and generally assists in the design and selection of facilities. The facilities which can be ordered from the telephone company follow.

- 2-wire trunks
  - TL 11M/E (E&M Type I signaling)
  - TL 12M/E (E&M Type II signaling)
- 4-wire trunks
  - TL 31M/E (Type I signaling)
  - TL 32M/E (Type II signaling)

Several qualities of conditioning beyond the "basic" line quality are available, including: C1, C2, C3, C4, and D1. Conditioning is the tolerance on frequency response and delay distortion. Generally, the "basic" line quality is adequate for voice applications. The conditioning is usually required for voiceband data applications.

Table 7
Private network facility requirements

From	To	Facility required	SL-1 cards
Node	Node	TL 31M/E interface (4-wire E&M Type 1)	QPC237
Node	Main	TL 31M/E interface (4-wire E&M Type 1)	QPC237 QPC71
Main	Satellite	TL 11M/E or TL 12M/E (2-wire E&M Type 1)	QPC71

When telco facilities are ordered, the FCC registration number must be provided. Different interfaces must be ordered for non-registered equipment. It is acceptable to have registered and non-registered equipment attached to the same facility, as well as to have a tie trunk between telco and customer-provided PBX equipment. When ordered from a telco, the facilities are furnished with VNL loss, or 0 dB loss if echo suppression is provided. When customer-owned facilities are used, it is the customer's responsibility to insert the VNL loss into the facility.

Public network facilities

Public network facilities must be ordered from the telephone company. A summary of these facilities is given in Table 8. The facilities must be identified as connecting to customer-provided equipment and the FCC registration number must be given. The facilities ordered can include:

– PBX central office trunks

– PBX foreign exchange trunks, to specific foreign exchanges

– PBX inward WATS trunks

– PBX outward WATS trunks

– off-premise stations

In addition, direct trunks to SPRINT and other common carrier systems may be ordered from these service suppliers. These services provide indirect access to the public network.

Table 8
Public network facility requirements

Service required	Facility required
Local calling area	PBX—C.O. trunk
Calling area local to a distant exchange	PBX FEX trunk
Wide calling area within the same state	Intra-state WATS outgoing PBX trunk
Calling area, all bordering states	Interstate WATS outgoing PBX trunk, band 1
Calling area within USA, Canada, bordering states and beyond	Interstate WATS outgoing PBX trunk, band 2 and higher
Calling to major cities in United States	Non-Bell services such as Sprint or MCI

Transmission performance

Voice quality performance

The quality of voice connection made over tandem trunks is a function of the composite characteristics of the trunks. Each trunk added to the connection degrades the overall transmission performance. Thus, some limits must be placed on the number of trunks permitted in tandem, as well as which trunks may be connected.

To maintain adequate voice quality while at the same time keeping the routing restrictions from becoming unduly complex, ESN is partitioned into two basic connection categories, each with its own set of requirements. The connection categories are:

- node-to-node
- node-to-main, satellite or tributary

Node-to-node connections

The restrictions on node-to-node connections are:

- No trunks have a loss exceeding 3.5 dB.
- A combination of trunks used for a valid connection not having a loss exceeding 4.1 dB.
- Split echo suppressors are provided at each end of each trunk equipped with echo suppressors. Each echo suppressor is enabled or disabled by the switch at its end.
- Tandem connections of echo-suppressor controlled trunks are permitted, provided the intermediate echo suppressors are disabled. The switch disables the echo suppressors it controls when a direct connection is made between two echo-suppressor controlled trunks, thus meeting this requirement. Routes with one (or more) intermediate non-echo-suppressor controlled trunks between echo-suppressor controlled trunks are not allowed because the intermediate echo suppressor is not disabled.
- Generally, no more than three tie trunks should be connected in tandem. A limit of four is imposed between echo suppressor controlled trunks. Software is arranged to disable echo suppressors when it tandems a call from an echo suppressor controlled trunk to another such trunk. It does not disable echo suppressors on other connections.

Node-to-main, satellite or tributary connections

Restrictions on these tie trunks follow.

- The node-to main trunk is normally a land circuit not exceeding 250 miles (400 km). If this objective cannot be met, the main PBX is treated as a node for transmission planning. The transmission planning of the node-to-main tie trunk is considered part of node-to-mode transmission planning.
- The node-to-main tie trunk, if less than 250 miles, must have a loss not exceeding 2.5 dB.
- Main-to-satellite and main-to-tributary trunks must have a loss not exceeding 2 dB.

Note: In some cases, the telco may only be able to provide non-VNL trunks which have a loss exceeding these objectives. If the loss is significantly higher than VNL, the switchable pad must be in the "pad-out" mode for connection to these trunks.

To minimize toll charges,.ESN can route calls over private network facilities to public network trunks. The public network has a designed loss which does not take into account the added loss of extending the call over private network facilities.

ESN, as any other private network, provides a lower quality performance on these connections than if the call were routed directly to the public network. The amount of degradation must be kept small enough that the connection is acceptable to most users. This loss is restricted as follows:

- Off-network long distance connections are only to be established from trunks terminating on nodes. The private network loss added to the public network loss is limited to:
  - 4.1 dB for calls originating at node stations
  - 6.6 dB for calls originating at main stations.
- Off-network local connections can exit at node and main PBX. For nodes, the loss is as above. For mains, the loss is limited to:
  - 6.6 dB for calls originating at node stations
  - 9.1 dB for calls originating at main stations.

Voiceband data performance

Voiceband data modems are used to transmit data between private network switches. General guidelines on the expected performance of various modem types for different ESN connections are provided in this document.

The guidelines are based on documented transmission performance of Bell System private lines and actual measurement of Meridian-1 and several metropolitan area private lines. The expected performance can be stated in statistical terms only to reflect the wide performance range of actual circuits.

The probability of success in completing a data call is based on an overall average of all Bell System private lines. A particular line or group of lines may be worse or better than the average. Therefore, the expected performance stated here should be viewed as a general indication only.

Tables 9 and 10 specify the expected performance for ESN connections with 1, 2, and 3 trunks in tandem. The percentage of successful calls is given for a given modem bit-rate. Table 9 applies to modems with Automatic Adaptive Equalizers and no significant improvement is expected with C2 or C1 conditioned lines relative to the basic quality private lines. Using D1 conditioned lines, (guaranteed lower noise) can improve performance as shown in the table. For modems without automatic (i.e., fixed) equalizers, significant improvement can result from using C1 or C2 conditioning over basic line performance as shown in Table 10.

Table 9
Performance of modems with automatic adaptive equalizers

Modem bit rate (probability of success)		
Number of trunks connected in tandem	**Basic C1, C2 conditioning**	**D1 conditioning (4-wire facility)**
1	2400 b/s (80%)	4800 b/s (<80%)
2	2400 b/s (50%)	4800 b/s (50%)
3	Not recommended	2400 b/s (50%)

Table 10
Performance of modems with fixed equalizers

Modem bit rate (Probability of success)			
Number of trunks connected in tandem	**Basic conditioning**	**C2 conditioning**	**C3 Conditioning**
1	2400 b/s (75%)	2400 b/s (80%)	2400 b/s (90%)
2	2400 b/s (50%)	2400 b/s (60%)	2400 b/s (75%)
3	Not recommended	2400 b/s (50%)	2400 b/s (50%)

Maintaining transmission performance

Maintaining high transmission performance in private networks requires constant vigilance on the part of the network administrators. The experience in the Northern Telecom network demonstrates that transmission performance degrades over time. The degradation can be traced to inadequate maintenance of the facility by the supplier and differences in performance when the supplier makes administrative changes in the assignment of equipment to provide tie trunk service.

Installation

After a tie trunk facility is installed, a transmission test is performed to verify that the facility meets tariffed requirements. Only after the performance has been verified should the facility be turned up for service.

Scheduled maintenance

Scheduled maintenance is performed at regular intervals whether or not trunk faults are known to exist. Testing should be carried out once a week until it is determined that less frequent testing is required.

Corrective maintenance

Follow-up to trouble reports generated by users or software diagnostics identifies various trunk problems in need of correction.

Transmission testing of tie trunks

ESN switches have been provided with the capability of accessing a quiet termination or 1020-Hz test tone at a remote ESN switch. These capabilities permit the testing of tie trunk transmission performance. Read *Automatic trunk maintenance feature description* (553-2751-104) for further information on trunk transmission testing capabilities.

Testing loss

The following test is performed at each end of each tie trunk so that both directions of transmission are checked:

1. From the maintenance terminal, load the trunk test program and access the remote test tone for the trunk to be tested.
2. Connect a transmission level meter to the "facility in" access jacks of the trunk under test. Measure the level of the 1020-Hz test tone. The permitted level requirements, based on a switched-in pad mode at the far-end switch, are given in Table 11.

Table 11
Transmission level requirements

Trunk design loss (dB)	Minimum (dBm)	Maximum (dBm)
0	–17.5	–11.5
0.5	–18	–12
1.0	–18.5	–12.5
1	–19	–13
2.0	–19.5	–13.5
2.5	–20	–14
3.0	–20.5	–14.5
3.5	–21	–15

If the requirement in Table 11 is not met, the fault must be isolated to the trunk facility or equipment. The following test aids in this isolation perform the test at each end of the facility.

1 From the maintenance terminal, load the trunk test program and access the local test tone.

2 Connect a transmission level meter to the "equipment " out jack of the suspect trunk. The level requirement is:

   - For QPC71 circuit packs: –15 dBm minimum, –13 dBm maximum
   - For QPC237 circuit packs: –15.5 dBm minimum, –13.5 dBm maximum

If the requirements in bullet two are not met at either or both facility ends, corrective maintenance must be performed and the test repeated. If the requirement is met at both facility ends, the facility supplier should be requested to perform corrective maintenance.

Testing noise

The following test is to be performed at each end of each tie trunk, so that both directions of transmission are checked.

1 From the maintenance terminal, load the trunk test program and access the remote quiet termination for the trunk to be tested.

2 Connect a noise meter to the "facility in" access jacks of the trunk under test. The noise requirements are given in Table 12.

Table 12
Transmission noise requirements

Distance miles (km)	Maintenance (dBrnC)	Immediate action (dBrnC)
0–15 (0–24)	28	36
16–50 (26–80)	28	36
51–100 (82–160)	29	36
101–200 (162–320)	31	36
201–400 (322–640)	33	40
401–1000 (642–1600)	35	40
1001–1500 (1602–2400)	36	40
1501–2500 (2402–4000)	39	44
2501–4000 (4002–6400)	41	46

Note: Trunks with a noise measurement within the MAINTENANCE category may be left in service. Trunks with a noise measurement within the IMMEDIATE ACTION category should be immediately removed from service. In either case maintenance action should be promptly initiated.

If the requirements of Table 12 are not met, the fault must be isolated to the trunk facility or equipment. The following test will accomplish this isolation. Perform the test at each end of the facility.

1. From the maintenance terminal, load the trunk test program and access the local quiet termination.
2. Connect a noise meter to the "facility out" jack of the suspect trunk. The requirement is that the noise not exceed 23 dBrnC.

If the requirement is not met at either or both facility ends, corrective maintenance should be performed on the appropriate Meridian 1 and the test repeated. If the requirement is met at both facility ends, the facility supplier should be requested to perform corrective maintenance.

Transmission considerations for remote network access

One application of private networks is to reduce toll charges on public network to public network calls. The user in the public network makes a local or toll-free (INWATS) call to one of the private network switches, then calls to an off-network destination via either Direct Inward System Access (DISA), or attendant assistance.

Gain

The public network transmission plan does not support tandems of two (or more) connections. Such tandems inherently results from using the private network to make a public network to public network call. The loss can be partially offset by gain.

Public network to public network calling via the private network has relatively small usage and savings, so it does not justify change to the private network transmission plan. Instead, gain is applied on the access trunks, which are used exclusively for incoming calls. The gain is applied independently of the connection established through the private network. Thus, the amount of gain provided is a compromise which optimizes grade of service on the more important connections, but possibly degrades service on some others.

Gain devices

The gain required is bidirectional between 2-wire interfaces and is provided by devices called repeaters. There are two types of suitable bidirectional gain devices: fixed and switched-gain. Fixed-gain devices are extremely sensitive to impedance mismatches at the 2-wire interfaces. Such mismatches can cause oscillation. Switched-gain overcomes the oscillation problem, but introduces speech impairments due to the switching action.

Fixed gain

Bi-directional fixed gain can be implemented either by an amplifier on the 2-wire path or two unidirectional amplifiers in a 4-wire arrangement, interfacing the 2-wire path through hybrids. Both schemes are sensitive to the impedances at the 2-wire interfaces. Adjustable matching networks are required to allow the device to interface a variety of facilities.

The more gain required, the closer the impedances must be matched to the interfaces in order to prevent oscillation or poor transmission associated with near oscillation. The impedance at the interfaces is partly determined by the impedances of facilities switched into a connection. Ultimately, those impedances limit the practical gain.

Voice-switched gain

The voice-switched gain amplifier avoids the stability problems by dynamically applying gain. Only one direction of transmission can have gain at a time. A loss equal to the gain is provided in the opposite transmission direction.

Gain is applied by monitoring for speech and applying gain in the talker-to-listener direction when speech is detected. In the idle mode, a small loss is inserted in both directions. Compensation for impedances is required in order for the direction sensing circuitry to function properly. Table 13 compares fixed gain and voice-switched gain.

Gain with compression

On some connections, there is the possibility that signal levels will already be high and gain may increase the levels above FCC specified limits. Some manufacturers provide compression options to guard against exceeding FCC limits. Speech compression ensures that signal levels will not exceed a specified maximum: –9 dBm generally. When speech levels, if amplified without compression, exceed that level, the gain is dynamically reduced so the output level does not exceed –9 dBm. The –9 dBm limit is specified in FCC requirements.

Gain with compression is possible on both fixed gain and voice-switched gain repeaters and does not change the adjustment procedure.

Table 13
Comparison of fixed gain and voice-switched gain

Fixed gain	Voice-switched gain
Does not degrade speech by gain switching action	Gain switching action may degrade speed.
Must permanently attach to a trunk and be adjusted for its impedance.	May be switched into connections as required, but is affected somewhat by facility impedance.
Not sensitive to voice levels.	May require voice activation adjustment to provide required sensitivity.
May oscillate under certain circumstances.	"Unconditionally" stable.
"Transparent" to voiceband data and DTMF signaling.	Must be switched out for voice-band data. May impair DTMF signaling.

Recommendation

When the required test equipment is available and the type of facility provided by the telco is known, fixed-gain units should be used because the performance is superior. However, switched-gain units should be used where the required transmission performance is not so critical, or the more complex alignment procedure for fixed-gain units cannot be performed.

Adjustment

The gain should initially be set to 6 dB in each transmission direction. This value of gain is a compromise between optimized grade of service and practical considerations to avoid oscillation and other transmission impairments. Based on experience, the gain may then be adjusted to greater or less than 6 dB.

Application to trunks

To minimize the effect on the overall transmission plan, the gain units should be installed only on trunks which are used to access the network primarily for calling via tie trunk facilities or to off-network destinations. Calls originated on-network are to be blocked from accessing these trunks.

Arrange one-way incoming DISA C.O. trunk groups, using ground start trunks. The trunks are used for calling over tie trunks and off-network. Calls terminating on the same PBX should be placed by the attendant or DID trunks, if provided.

When DISA is equipped on DID trunks, do not provide gain units. DID trunks carry traffic both to telephones and DISA, and gain is not desirable for the traffic to telephones. Instead, a new ground-start C.O. trunk group should be added and DISA moved to that group. Gain units can then be provided on that group.

Glossary

Following are definitions of the terms used within this publication.

Public network facilities

PBX Central Office trunks

PBX Central Office (CO) trunks connect the PBX switch to the central office which serves the PBX location. The trunks appear as station lines at the central office equipment.

Foreign Exchange Trunk

A Foreign Exchange (FEX) trunk provides a direct connection between a PBX switch and a remote central office other than the central office which serves the location of the PBX.

Wide Area Telecommunications Service

Wide Area Telecommunications Service (WATS) provides a bulk rate service for incoming or outgoing toll calls within selected geographical regions (bands). A WATS trunk must terminate on a central office equipped to provide WATS, which may be the office that would normally serve that customer, or it may be a remote central office.

– Outgoing WATS

  An outgoing WATS (OUTWATS) trunk is used exclusively for outgoing bulk-rate calls from a customer's PBX to a defined geographical area by the toll network.

– Incoming WATS

  An incoming WATS (INWATS) trunk is used exclusively for incoming calls from a defined geographical area to a customer's PBX.

2-wire facility

A 2-wire facility is characterized by supporting transmission in two directions simultaneously, where the only method of separating the two signals is by the propagation directions. Impedance mismatches cause signal energy passing in each direction to mix with the signal passing in the opposite direction.

4-wire facility

A 4-wire facility supports transmission in two directions, but isolates the signals by frequency division, time division, space division, or other techniques that enable reflections to occur without causing the signals to mix together. A facility is also called 4-wire if its interfaces to other equipment meet this 4-wire criteria (even if 2-wire facilities are used internally) as long as crosstalk between the two transmission directions is negligible as measured at the interface.

PBX types

Main PBX

A main PBX is one which has a Directory Number (DN) and can connect PBX stations to the public network for both incoming and outgoing calls. A main PBX can have an associated satellite PBX. A main PBX can be part of a Tandem Tie Trunk Network (TTTN). If the main PBX provides tandem switching for tie trunks, it is called a tandem PBX. In the context of ESN, a main PBX has tie trunks to only one node.

Satellite PBX

A satellite PBX has no direct incoming connection from the public network. All incoming calls are routed from an associated main PBX over tie trunks. This definition places no restrictions on the handling of outgoing calls from the satellite PBX. A satellite PBX can have one-way outgoing trunks to the central office, in addition to outgoing service on trunks to the main PBX. In the context of ESN, a satellite PBX has no direct trunks to a node; however, calls to the node can be made through the main PBX.

Tributary PBX

The only difference between a satellite and tributary PBX is that the tributary PBX has direct incoming connection from the public network.

Tandem PBX

A tandem PBX is used as an intermediate switching point in a TTTN to connect tie trunks together, in addition to the usual PBX functions.

ESN node PBX

An ESN node PBX is a Meridian 1 switch equipped with the node software package of ESN. It performs tandem switching with software-controlled alternate route selection to bypass busy trunk groups.

ESN main PBX

An ESN main PBX is a Meridian 1 switch with the ESN main software package. It performs tandem switching between a node and the main's satellite and tributary switches. An ESN main PBX has outgoing tie trunks to only one node.

PBX networks

Tandem Tie Trunk Network

A Tandem Tie Trunk Network (TTTN) is a switched customer network which uses tie trunks to interconnect PBX switches in different locations. Calls are routed between the switches by progressive dialing of access codes.

Common Control Switching Arrangements

A Common Control Switching Arrangement (CCSA) is a Bell system offering which divides equipment between PBX switches and tandem switches. The tandem switches are located on telco premises, and perform only a tandem switching function. Network calls by a user at a PBX switch are made by dialing an access code (typically the digit "8") which connects the user to the telco tandem switch. The user then dials a 7-digit number to reach the desired telephone at another connected switch. Routing between tandem switches is performed under common control.

Electronic Switched Network

An Electronic Switched Network (ESN) is a Northern Telecom private network offering which has a dialing plan similar to that used in a CCSA. However, tandem switching functions are performed by PBX switches located on the customer's premises. Fewer restrictions on tie trunk routes are required by ESN than by the CCSA. CCSA requires that at least two tie trunks be connected in tandem to route a call between a pair of PBX switches. ESN uses one tie trunk to perform the same routing wherever there is sufficient traffic to justify the route.

Transmission Level Point

As an analog signal passes over a transmission facility, it encounters gains and losses which are part of the facility design and net out to zero. For example, a facility which uses a cable may have loss in the cable and may compensate for this loss by gain in one or more amplifiers. On the other hand, some losses are introduced to improve grade of service and are not recovered. The term Transmission Level Point (TLP) is used to discriminate a gain or loss which is recovered from one which is not.

TLP requires definition of a point in a transmission system as a reference level. Once this point is defined, the reference levels at other points can be derived. For example, the input to a transmission system might be defined as a –2 TLP. Assume that the signal passes through a 10 dB amplifier which compensates for loss elsewhere. The output of the amplifier is at a +8 TLP([–2] + [+10]). A –10 dBm signal at the input will be 0 dBm at the output. The same signal is –8 dBm0 (8 dB below the reference level) at the system input. It is also –8 dBm0 at the amplifier output. The signal has the same level, referenced by the TLP, because the gain in this case is offset by loss elsewhere, and does not show up as gain or loss at the signal destination. A second example is a transmission system with a 2 dB deliberate loss. The input and output TLPs are –2. A signal in at –10 dBM is –8 dBm0. The signal leaves the system with a level of –12 dBm, or –10 dBm0. The difference in level, even though the TLP is the same, is the loss introduced deliberately in the facility.

2-wire and 4-wire trunks in PBX networks are usually –2 TLP. However, some 4-wire trunks have a –16 TLP at the facility input and a +7 TLP at the facility output. If the trunk has a 0 dB loss, an input signal of –26 dBm (which is –10 dBm0), is received at a level of –3 dBm (23 dB gain) which is also –10 dBm0. This means that loss elsewhere in the connection cancels out the 23 dB in the facility.

SL-1

Electronic Switched Network

Transmission guidelines

Release 2.0
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SL-1

Basic and Network Alternate Route Selection

Description

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This document is issued to include updates and changes for X11 release 16. All updates are noted with revision bars in the margins.

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This document is reissued to include technical content updates. Due to the extent of changes revision bars are omitted.

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This document is reissued to include information on system option 81 and equipment required for compatibility with X11 release 18. Only new information and changes to technical content are noted by revision bars in the text.

August 1, 1993

This document is reissued for updates and changes resulting from X11 release 19. All updates are noted with revision bars in the margins.

October 31, 1993

This document accompanies the product release that includes North American Numbering Plan changes.

Contents

List of figures

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Introduction

The Basic Alternate Route Selection (BARS) and Network Alternate Route Selection (NARS) features enable a customer with a number of Meridian 1 systems in different locations to create a private telecommunications network (shown in Figure 1). The BARS or NARS features direct a call from a Meridian 1 in one geographical location to a Meridian 1 in any other geographical location in a cost-efficient and easy-to-use manner by:

- eliminating long, complex dialing plans and replacing them with an abbreviated Uniform Dialing Plan (UDP) common to all Meridian 1 that are part of the network
- providing a means of controlling the number and types of trunks that are available to each network caller and a method of controlling the time of day that access to a trunk (or group of trunks) is allowed
- automatically selecting the least-cost trunk route available to complete a call between network switches
- providing uniform network access to stations served directly at a Meridian 1 Node and stations served at Meridian 1 Mains or Conventional Mains connected to a Meridian 1 Node by tie trunks
- providing the call originator with the option to accept or refuse call completion over an expensive trunk, if less expensive trunks are not currently available
- providing optional queuing features that enable a call originator (when all trunks are busy) to remain off hook until a trunk becomes idle or hang up and receive a call back from the Meridian 1 when a trunk becomes idle.

Whether the BARS or NARS feature is used depends on the number and placement of Meridian 1 within the customer's private network, as well as the volume of traffic the network is to carry. NARS has the ability to translate location codes while BARS translates codes used for on-net dialing as Special Numbers. Refer to the section describing on-net to off-net automatic overflow.

When used in large, widely dispersed applications, NARS forms an integral part of Northern Telecom's Electronic Switched Network (ESN) product. For more complete details, refer to *Electronic Switched Network description* (309-3001-100).

Document overview

This document describes BARS/NARS, including the following elements:

- Network access codes
- Uniform Dialing Plan (UDP)
- Dialing transparency
- Automatic least-cost routing
- Digit manipulation
- Time of Day (TOD) routing
- Routing control
- Network controls:
  - Network Class of Service (NCOS)
  - Traveling Class of Service (TCOS)
  - Facility Restriction Level (FRL)
  - Class of Service (CLS)
- Digit manipulation, translation, restriction, and recognition:
  - 1-4 digit translation and 1-7 digit restriction (X11 release 4)
  - 1-4 digit translation, 1-7 digit restriction, and 1-7 digit recognition (X11 release 5)
  - 1-11 digit translation, restriction, and recognition (X11 release 8)

- Free Calling Area Screening (FCAS)
- Expensive Route Warning Tone (ERWT)

It also includes sections on implementing BARS and NARS.

Terminology

Terminology used in this publication is based on the following definition of switch types:

Meridian 1 ESN Node
A Meridian 1 equipped with the NARS or BARS feature.

Meridian 1 ESN Main
A Meridian 1 connected by tie trunks to a Meridian 1 Node and equipped with the Network Signaling (NSIG) feature package. The connected Meridian 1 Node must also be equipped with the NSIG feature package.

Conventional Main
A Meridian 1 switch connected by tie trunks to a Meridian 1 Node, but not equipped with the Network Signaling feature package. Conventional Main also applies to any other switch type (for example, step-by-step) connected by tie trunks to a Meridian 1 Node.

Other documentation

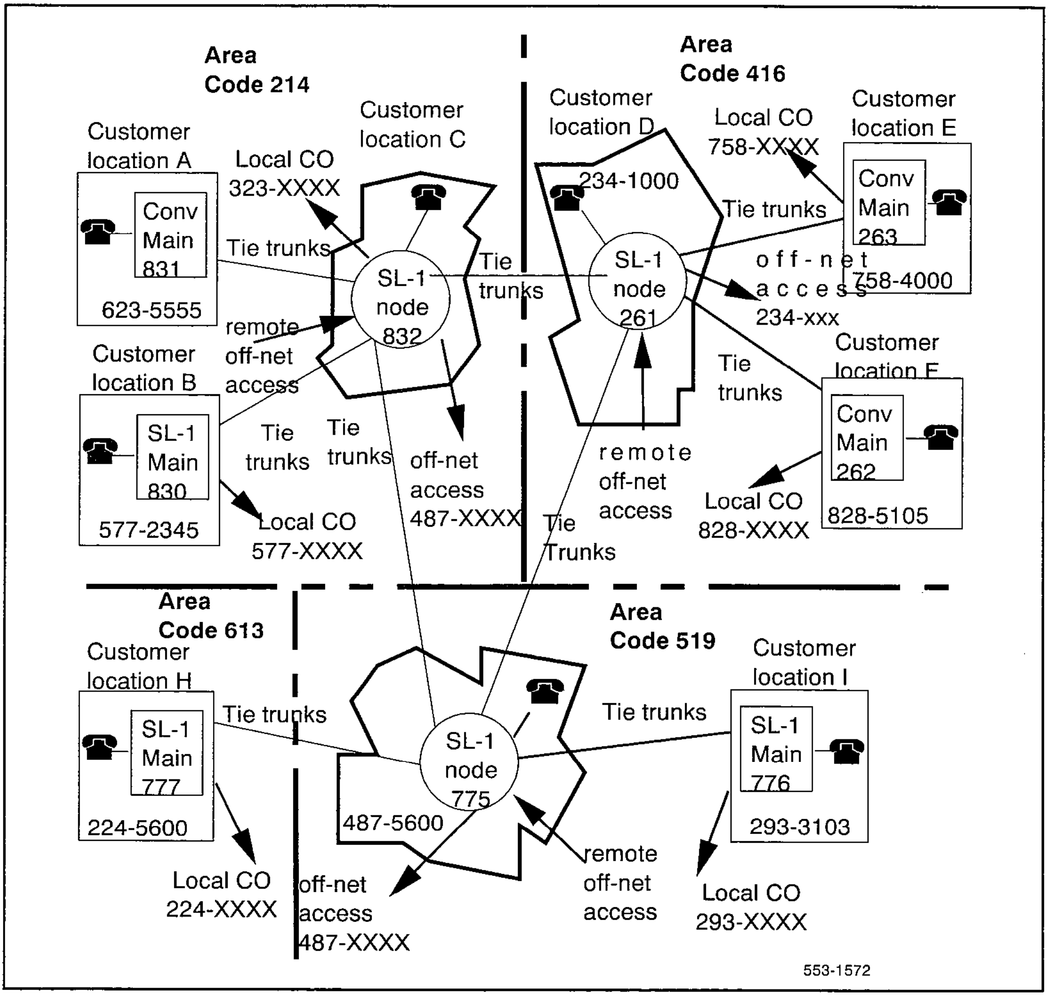
Other optional features that enhance the capabilities of BARS or NARS (such as Off-Hook Queuing and Network Authorization Codes) are mentioned briefly in this publication. The following publications describe these optional features:

Network Queue description (553-2751-101)

Coordinated Dialing Plan description (553-2751-102)

Basic and Network Authorization Code description (553-2751-103)

Figure 1
A typical private telecommunications network



Basic/Network Alternate Route Selection

The BARS and NARS features provide comprehensive and flexible networking packages that can be configured to satisfy the specific requirements of a customer's private network. The BARS or NARS packages provide benefits to users through simplified dialing plans and reduced communications costs. This section describes the prime elements of the BARS and NARS features.

Access codes

By dialing a one- or two-digit access code, the user can place a long distance, network, or local call. These BARS and NARS access codes are customer-defined.

BARS access code

To make long distance calls and calls to distant company locations (on-network calls) from the Meridian 1 Node, the caller dials the BARS access code (AC1) followed by the desired number. The BARS access code can be any one- or two-digit number, provided there is no conflict with any other part of the dialing plan. It is a customer-defined option whether the caller hears the dial tone after dialing a BARS access code.

Dialing the BARS access code triggers the BARS software to perform the call processing and routing required for call completion. This is accomplished by means of a network translation table associated with the BARS access code. Normal translation is used for all other call types.

NARS access codes

To access NARS, the user dials either of the two customer-defined NARS access codes: AC1 (for on-net and long distance calls) and AC2 (for off-net and local calls). Typically, AC1 is 8 and AC2 is 9. However, any one- or two-digit codes can be used, provided that AC1 is different from AC2 and there is no conflict with any other part of the dialing plan. It is a customer-defined option whether the caller hears the dial tone after dialing a NARS access code.

Dialing a NARS access code triggers the NARS software to perform the call processing and routing required for call completion. This is accomplished by means of Network Translation Tables. There is a Network Translation Table associated with each NARS access code. This translation mechanism is used to implement the NARS uniform dialing plan for private networks.

Network translation

The normal digit translator reads the dialed network access code, determines if the call is to be processed by BARS or NARS, and selects the appropriate Network Translation (Table 1). BARS or NARS translation determines the method to be used to process the call, refers to Supplementary Digit Restriction and Recognition tables if required, and applies digit restriction or recognition where it is specified. The result of translation is to invoke either route selection with a specified route list, standard call blocking, queuing, or internal recognition.

Table 1
Digit translation

Type	Before X11 release 8	X11 release 8 and later
LOC	3	3–7
HLOC	3	3–7
NPA	3–4	3–11
HNPA	3–4	3–11
NXX	3–4	3–8
SPN	1–4	1–11

Any Meridian 1 running a version of X11 software prior to X11 release 8 will translate the first three digits after the access code – first four digits with 1+ dialing or four-digit SPN (Special Number) codes. (SPNs can be one to four digits.)

Each NPA, NXX, SPN, or LOC associates with only one route list index. In a network with multiple switches sharing DID numbers within the same NXX, Coordinated Dialing Plan is needed to route the calls properly.

11-digit BARS or NARS translation

With 11-digit translation introduced in X11 release 8, the ESN BARS or NARS translation capabilities are expanded from 4 digits to a maximum of 11 digits for route selection. This means more digits are translated and therefore make similar codes translate differently. This increases the flexibility for network routing, including international calls. For example, NXX1363 becomes 13634 or 13635. More than one code can be handled by the same route list.

By allowing translation of more than four leading digits, unique non-conflicting routing to a destination is possible. More than one route list can exist for each specific code of a type. For example, the NXX 727 could only translate into one route list previously. With 11-digit translation, as many route lists as are needed to eliminate code conflict or achieve network requirements can be defined by extending translation deeper into the dialed code. Table 1 compares the number of digits that can be translated prior to X11 release 8 with present capabilities.

Uniform Dialing Plan

The Uniform Dialing Plan (UDP) is not a feature of BARS or NARS, but is an operation that enables users at a Meridian 1 Node, Meridian 1 Main, or Conventional Main to dial all calls in a uniform manner regardless of the location of the calling party or the route that the call will take. UDP applies to BARS on X11 release 5 using off-net number recognition and calling Home Location Code (HLOC) and SPN or NXX codes. LOCs are programmed as NXXs or SPNs.

UDP for on-net calling

An on-net call is one that terminates at a customer-owned location. To reach any on-net location, the user dials the NARS or BARS on-net access code (AC1), followed by seven digits. The dialing format for this call would be:

AC1 * LOC + XXXX (For NARS)

AC1 * SPN + XXXX (For BARS)

Legend:

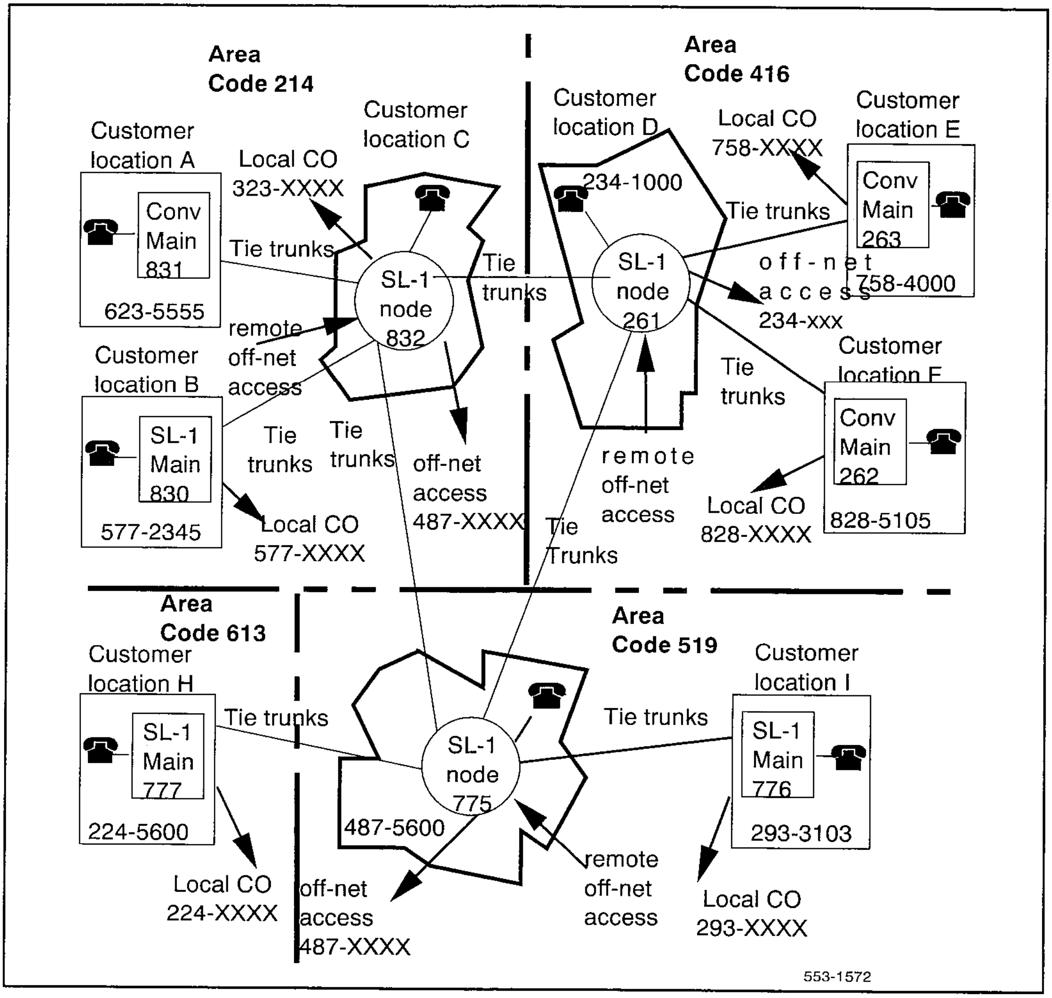
AC1	=	the one-digit or two-digit BARS/NARS on-net access code
*	=	NARS dial tone (optional)
LOC	=	a three-digit location code assigned for the destination location
SPN	=	a three-digit code assigned for the destination location
XXXX	=	the extension number of the party to be reached at the destination location

Each switch that is part of the BARS or NARS network is referenced by a unique three-digit location (LOC), SPN, or NXX code assigned at the Meridian 1 Node. There must be no conflict between the Location Code number assigned for a switch and Number Plan Area (NPA) codes, NXX, or other SPN.

A customer-owned location can be either physically connected to the network (by means of private trunk facilities) or virtually connected to the network (by means of public facilities). If a location is virtually connected to the network, the dialed LOC code is translated and converted by the Meridian 1 Node into the public number for the virtual location; for example, the Direct Distance Dialing (DDD) number or the direct inward dial (DID) number. Using digit manipulation for BARS, an SPN can be changed into the public network LDN.

Figure 2 illustrates a private network with a typical UDP. A user at LOC 830 (customer location B) calling extension number 3283 at LOC 777 (customer location H) would first dial 8 (AC1), pause for NARS dial tone (optional), then dial 777-3283. A user at any other customer location that is part of the network would dial these same numbers to reach extension 3283.

Figure 2
Example of a network with a typical NARS Uniform Dialing Plan



UDP for off-net calling

An off-net call is one that does not terminate at a customer-owned location, even though some on-net facilities may be used to complete a portion of the call routing. Referring to Figure 2, a call would be termed off-net if a user at LOC 776 called a station number associated with CO 758-XXXX in the foreign area code 416.

Tables 2 and 3 list the dialing formats for the various types of NARS and BARS UDP calls respectively.

Table 2
Dialing formats for NARS UDP calls (Part 1 of 2)

Call type	Dialing format	Code capacity
On-Net (Notes 1, 2, and 3)	AC1 * LOC + XXXX	640
DDD FNPA (Notes 1 and 4)	AC1 * 1 + NPA + NXX + XXXX	160
Network Speed Call	AC1 * LA + LN	1 - nn
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX	160
International DDD	AC1 * 011 + CC + NN	99
Operator-assisted International DDD	AC1 * 01 + CC + NN	99
DDD HNPA (Note 1)	AC1 or AC2 * 1 + NXX + XXXX	1
DDD operator	AC2 * 0	1
Local calls (Note 1)	AC2 * NXX + XXXX	640
Special local services	AC2 * SPN	8
Toll-free calls	AC2 * 800 + NXX + XXXX	1
Toll-free calls (Note 1)	AC2 * 1 + 800 + NXX + XXXX	1
Toll Special Numbers	AC2 * 900 + NXX + XXXX	1
Toll Special Numbers (Note 1)	AC2 * 1 + 900 + NXX + XXXX	1

Table 2
Dialing formats for NARS UDP calls (Part 2 of 2)

Note 1: If 1+ dialing is used, the on-net and local calls code capacities are increased to 800 and 792 respectively.

Note 2: If the code 1XX is reserved for future 1+ dialing use, and not for Network Speed Call codes, then the Location Code capacity will be reduced to 639 if a three-digit Network Speed Call (NSC) code is used, 632 if a two-digit NSC code is used, or 560 if a one-digit NSC code is used.

Note 3: When 1+ dialing is used, Network Speed Call access will be in the form of 2XX-9XX as a subset of the location codes utilized in the UDP. The Location Code capacity will be reduced to 799 if a three-digit NSC code is used, 792 if a two-digit NSC code is used, or 720 if a one-digit NSC code is used.

Note 4: See this page and following page for legend of acronyms and dialing format abbreviations.

Legend

AC1	Access code for on-net, long distance and Network Speed Calls. Typically the digit "8," but can be either one or two digits in length.
AC2	Access code for local calls. Typically the digit "9," but can be either one or two digits in length.
*	Symbol meaning NARS dial tone (optional)
NPA	Numbering Plan Area (NPA) code. Any number of the form NXX.
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form NXX.
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form NXX.
N	Any of the digits from 2 to 9.
X	Any of the digits from 0 to 9.
CC	Country code. Any one, two, or three digits from 2 to 9.
NN	National Number. Depends on national dialing plan; maximum 12 digits including the country code.
LA	List access code. Any one, two, or three digits from 0 to 9.
LN	List element number. Any one, two, or three digits(0-9, 00-99, 000-999).
LOC	Three-digit Location Code for each UDP network location.
NXX	Local Exchange Code.
XXXX	Four-digit directory (extension) number.
SPN	Special Numbers, for example 411, 611, or may be XXXX.

Table 3
Dialing formats for BARS calls (Part 1 of 2)

Call type	Dialing format
On-Net	(1) AC1 * SPN + XXXX (2) AC1 * NXX + XXXX
DDD FNPA	AC1 * 1 + NPA + NXX + XXXX
Operator-assisted DDD	AC1 * 0 + NPA + NXX + XXXX
International DDD	AC1 * 011 + CC + NN
Operator-assisted international DDD	AC1 * 01 + CC + NN
DDD HNPA (if applicable)	(1) AC1 * 1 + NXX + XXXX (2) AC1 * 1 + NPA + NXX-XXXX (see note)
DDD operator	AC1 * 0
Local calls	AC1 * NXX + XXXX
Special local services	AC1 * 411, 611, etc.
Toll-free calls	AC1 * 1 + 800 or 1 + 900 + NXX + XXXX

Note: 1 + dialing is optional; you can simply enter *1 NPA.*

Legend

AC1	Access code for on-net, long distance and Network Speed Calls. Typically the digit "9" but can be either one or two digits in length.
*	Symbol meaning wait for BARS dial tone (optional).
NPA	Numbering Plan Area (NPA) code. Any number of the form NXX.
HNPA	Home Numbering Plan Area (HNPA) code. Any number of the form NXX.
FNPA	Foreign Numbering Plan Area (FNPA) code. Any number of the form NXX.
CC	Country code. Any one, two, or three digits from 2 to 9.
SPN	Special Number. Code used to identify other office locations on the network, or a number such as 411 or 611.

Table 3
Dialing formats for BARS calls (Part 2 of 2)

Call type	Dialing format
NN	National Number. Depends on national dialing plan; maximum 12 digits including the country code.
N	Any of the digits from 2 to 9.
X	Any of the digits from 0 to 9.

Dialing transparency

Extending network access to a Meridian 1 Main or Conventional Main is accomplished by forming a single tie trunk access group from the main to the node. Users at main switches access the trunk group to the node by dialing the BARS or NARS on-net access code (AC1). The node is arranged to insert the digit(s) for AC1 on each incoming call from the main, thus enabling access to the network facilities of the node in a transparent fashion. Local calling is arranged through Conventional dial "9" CO trunks at the main.

> *Note:* If a Meridian 1 Node replaces a tandem switch in a tandem tie trunk network (TTTN), other tandem switches in the network can "tandem through" the Meridian 1 Node using the same access codes as before. This requires that there be no conflicts between the access codes for the TTTN trunks and the dialing plan implemented at the SL-1 Node.

Automatic least-cost routing

For each network call translated at a Meridian 1 Node, BARS or NARS selects a route from a list of outgoing alternate routes to complete the call. A list of alternate routes to a particular destination is called a route list and each route specified in the list is termed an entry. Any combination of trunks, including central office (CO), Foreign Exchange (FX), and tie, can be specified in a route list.

Note 1: Prior to X11 release 13, a route list can contain up to eight entries. With X11 release 13 and later, the route list can contain up to 32 entries.

Note 2: Use of the Bandwidth Controller (DCA System 9000) improves the capacity of T1-based tandem networks. Its dynamic alternate routing capability can independently choose the optimum path for a voice or data call. However, it is necessary to coordinate the Bandwidth Controller routing lists with the ESN routing lists.

Typically, the first entries (routes) in a route list are the less expensive routes to a destination and comprise the initial set (I set) of routes in the list. The remaining routes in the list (if any) are the more expensive routes to a destination and comprise the extended set of routes in the list. An initial set marker, defined through service change, determines which routes comprise the initial route set. Refer to *Electronic Switched Network description* (309-3001-100) for more information on I set and extended set routes.

A Meridian 1 Node equipped with NARS supports 256 route lists (0-255). A Node equipped with BARS supports 128 route lists (0-127).

Each route list entry has the following information defined:

– the route number
– the minimum FRL required for access
– the time of day the route can be accessed
– whether Off-Hook Queuing (OHQ) or Call Back Queuing (CBQ) is allowed on the route
– whether the route is to receive the Expensive Route Warning Tone (ERWT) treatment

- a digit manipulation table index number
- a Free Calling Area Screening (FCAS) table index number
- whether converting on-net numbers to off-net numbers is allowed (NARS)

Route eligibility

BARS or NARS translates the number dialed (1-11 digits) after an access code into a route list, and searches sequentially the routes in the list for an available route. Route eligibility for a given call is based on the caller's Network Class of Service (NCOS), the NCOS-defined FRL, the current time of day, and the originator's Class of Service (CLS).

Because each entry in a route list has a minimum FRL required for access and all network users are assigned an FRL through their NCOS, the network communications manager can restrict the type of calls allowed to particular users. For example, if the minimum FRL for all calls is 1, except for special local services numbers that are assigned an FRL of 0, a user assigned to an NCOS group with an FRL of 0 would only be able to make calls to the Special Numbers. In addition, the communications manager can restrict the use of high-cost facilities by assigning a high FRL to the expensive routes in a route list and a lower FRL to a user's NCOS.

Digit manipulation

Any trunk type can be specified in a route list. However, when certain trunk types are accessed, the digits dialed by the user must be manipulated to conform to the dialing requirements of the trunk. To do this, BARS or NARS uses digit manipulation tables to modify the dialed digits. There can be a maximum of 256 digit manipulation tables, each referenced by a Digit Manipulation Index number, defined at each Meridian 1 Node. Digit manipulation can delete up to 15 leading digits and insert up to 24 leading digits.

A user at customer location I (Figure 2) dials 8-613-596-9084 to reach an off-net station in the 613 NPA associated with customer location H. At the Meridian 1 Node, BARS or NARS selects the appropriate route list for call completion to NPA 613 and finds that the only available route to that NPA is a local CO trunk that requires the insertion of the leading digit "1" for long distance calls. The route list entry for this route specifies a Digit Manipulation Index number (0-255). "0" means no digit manipulation is required. BARS or NARS refers to the digit manipulation table indicated by the index number, deletes digits as specified in the table (none in this case), inserts the required digits ("1" in this case), and completes the call on this route.

Time of Day routing

BARS or NARS provides for up to eight Time of Day (TOD) schedules, numbered from 0 through 7. Each entry (route) in a route list is assigned to a TOD schedule that specifies the hour(s) during which the particular entry can be accessed. Thus, based on the current time of day, the most cost-effective route alternatives can be specified. A typical TOD schedule is shown in Table 4.

Based on the TOD schedule shown in Table 4, a route list entry assigned to TOD schedule 2 would be accessed only between the hours of 00:00 to 07:44 and 17:30 to 23:59. Access to the route at any other time would be denied. TOD schedules can be turned on or off through service change, as traffic conditions warrant. A TOD schedule is turned on for an entry by turning off all other TOD schedules. An X preceding the schedule number turns that number off. For example, for TOD schedule 2 to run, schedules 1 and 3 must be off.

Table 4
A typical TOD schedule

TOD schedule	Time period
2	00:00 to 07:44 17:30 to 23:59
1	07:45 to 08:59 12:00 to 13:14 16:00 to 17:29
0	09:00 to 11:59 13:15 to 15:59

Note 1: A TOD schedule can be associated with any number of arbitrarily selected 15-minute periods. However, any one 15-minute period can appear only in one TOD schedule.

Note 2: Normally, the user does not define 0, but lets the Meridian 1 define it by calculating times of day not mentioned in schedules 1-7.

Routing Control

The Routing Control feature provides a mechanism for changing a user's network-access capabilities when:

- a special TOD schedule is in effect
- an extended TOD is in effect
- the user presses a Routing Control key on the console.

NCOS map

With the NARS or BARS feature, TOD schedule 7 is the special TOD schedule. Associated with the special TOD schedule is a Network Class of Service (NCOS) map. The NCOS map lists all NCOS numbers. Associated with each listed NCOS is an alternate NCOS number (greater than, equal to, or smaller than) that replaces the original NCOS number when the special TOD schedule is in effect. Table 5 illustrates a typical NCOS map.

Table 5
A typical NCOS map for Routing Control

Original NCOS	Alternate NCOS (Note)	Original NCOS	Alternate NCOS (Note)
0	0	8	2
1	0	9	3
2	0	10	3
3	1	11	4
4	1	12	4
5	2	13	5
6	2	14	5
7	2	15	5

Note: The alternate NCOS replaces the user's original NCOS when Routing Control is in effect, due to TOD 7, Extended Time of Day, or when manually established at the console.

Invoking Routing Control

The alternate NCOS numbers associated with special TOD schedule 7 are normally invoked when the time specified for TOD schedule 7 corresponds to the time in the system clock. Additionally, the alternate NCOS numbers can be scheduled for implementation (through service change) for the full 24-hour period of specified days of the week. This capability enables network-access capabilities to be changed automatically on weekends or company holidays.

The attendant can also manually invoke the special TOD schedule through use of a Routing Control (RTC) key on the console. Pressing the RTC key lights the associated lamp, and invokes the special TOD schedule. To deactivate Routing Control, the RTC key is pressed again. The associated lamp goes dark, and normal TOD schedules are once again in effect.

> *Note:* Authorization Code can be used to override the restrictions imposed through Routing Control. If a user enters a valid Authorization Code (AUTH), the NCOS number associated with the AUTH is applied for the duration of the call. Routing Control is not affected.

Network Class of Service

Network Class of Service (NCOS) is an integral part of the BARS and NARS features at a Meridian 1 Node, and of the Network Signaling feature at a Meridian 1 ESN Main.

NCOS provides the means to control:

- which trunk routes are eligible for attempted call completion
- whether queuing is offered to the call originator
- whether the call originator receives a warning tone when an expensive trunk is selected to complete a call
- whether the user is allowed to access the Network Speed Call feature.

After NCOS groups are defined through service change, each line, trunk, and attendant group is assigned to the NCOS group that best serves its requirements. The NCOS group to which each line, trunk, or attendant group is assigned is independent of the assigned class of service. Tie trunks incoming from Meridian 1 Mains or Conventional Mains are also assigned to an NCOS group (at the Node), which determines their level of access to the network facilities at the Node. See *X11 features and services* (553-3001-305), and *X11 input/output guide* (553-3001-400) for more information.

Table 6 summarizes the NCOS and other parameters for Meridian 1 Nodes and Meridian 1 Mains.

Table 6
Summary of node and main parameters

Parameter (Note 4)	BARS Node	NARS Node	ESN Main
NCOS groups (Note 1)	0–99 (0–7)	0–99 (0–15)	0–99 (0–15)
Facility Restriction Levels	0–7	0–7	0–7
Digit manipulation tables	0–256	0–256	—
Route lists	0–127	0–256	—
Route list entries	0–31 (0–7)	0–31 (0–7)	
FCAS tables (Note 2)	1–127	1–255	—
SDR tables (Note 2)	0–255	0–511	—
SDRR^{5+} tables (Note 2)	0–255	0–512	—
TOD schedules (Note 2)	0–7	0–7	—

Note 1: The values in parentheses ( ) are for releases prior to X11 release 13.

Note 2: FCAS= Free Calling Area Screening
SDR= Supplemental Digit Restriction
SDRR= Supplemental Digit Restriction/Recognition
TOD= Time of Day

Note 3: If the NARS and BARS features are equipped in the same switch but for different customers, the highest parameter values apply to that switch. For example, if one customer has NARS and another customer has BARS, the NARS parameters apply to the BARS customer.

Note 4: If the New Flexible Code Restriction (NFCR) feature is equipped in conjunction with BARS or CDP, the number of available NCOSs is 100. Prior to X11 release 13, only 8 NCOSs are allowed. NFCR is described in *X11 features and services* (553-3001-305).

Network Control

Network Control is an enhancement to the Network Class of Service (NCOS) feature that extends NCOS controls to users located at a Meridian 1 Main. Network Control requires that the Meridian 1 Main and serving Meridian 1 Node be equipped with the NSIG feature. The Meridian 1 Main must also be equipped with the NCOS feature.

Meridian 1 Main NCOS

Users (lines, trunks, and attendants) at a Meridian 1 Main are assigned an NCOS that determines their level of access to network facilities at the serving Meridian 1 Node. When a user at a Meridian 1 Main initiates a call to (or through) a Meridian 1 Node, the user's assigned NCOS or TCOS, depending on tie trunk settings (ESN or ESN2, for example), can be transmitted.

Only NCOS 0 through 7 can be assigned at the node if the user's NCOS is transmitted to the serving node and the node is equipped with BARS. Therefore, only NCOS 0 through 7 should be assigned at the ESN Main. An ESN Main equipped with NSIG supports an NCOS of 0 through 99. If the node is equipped with NARS, an NCOS of 0 through 99 can be assigned at both the node and the ESN Main. Prior to X11 release 13, an NCOS of 0 through 15 is supported.

The transmitted NCOS or TCOS overrides the NCOS or FRL assigned to the incoming tie trunk group at the node, and is used to determine the user's eligibility for network resources/features at the Meridian 1 Node. Thus, a user at a Meridian 1 Main has the same network-access capabilities as a user at the Meridian 1 Node who is assigned the same NCOS.

> *Note:* If the user at the Meridian 1 Main enters a valid Authorization Code prior to placing a BARS or NARS call, the NCOS associated with the Authorization Code is transmitted to the Meridian 1 Node in place of the user's assigned NCOS.

Calls from a Conventional Main to the Meridian 1 Node are controlled by the NCOS assigned to the incoming trunk group at the Meridian 1 Node, as the Conventional Main has no NSIG.

Meridian 1 Node TCOS

Network Control at a Meridian 1 Node can provide a Traveling Class of Service (TCOS) mechanism. TCOS controls route access and Off-Hook Queuing (OHQ) eligibility for calls placed to (or through) another Meridian 1 Node or an associated Meridian 1 Main. It also enables the Meridian 1 Node to interface with switches that are part of an Electronic Tie Network (ETN) as long as the SIGO setting on the tie trunk route is set for ETN at both ends. Nodes can send NCOS if SIGO is set for ESN2, for example, at both ends of the tie trunk group.

The Traveling Class of Service is, in effect, the Facility Restrictions Level (FRL) of a user's assigned NCOS. When a user at a Meridian 1 Node initiates a call to another Meridian 1 Node (or a Meridian 1 ESN Main), the TCOS (for example, the FRL of the user's assigned NCOS) is transmitted to the other Meridian 1 Node. At the receiving Meridian 1 Node, the TCOS (0-7) replaces the FRL of the NCOS assigned to the incoming trunk group. Route access and OHQ eligibility for the call are, therefore, based on the NCOS of the incoming trunk group with the modified FRL (for example, TCOS).

> *Note:* The Network Control (NCTL) data block (LD87) is used to define OHQ eligibility on a per FRL (TCOS) basis. For example, if FRL 4 is defined as OHQ eligible, then all users with an NCOS with an FRL of 4 are eligible for OHQ on calls placed to another Meridian 1 Node or to an associated Meridian 1 Main.

If a user at a Meridian 1 Main or Conventional Main initiates a call that tandems through the serving Meridian 1 Node to another Meridian 1 Node or Meridian 1 Main, TCOS applies to the call as if the call originated at the serving Meridian 1 Node.

Compatibility with ETN switches

The Meridian 1 TCOS is equivalent to the Traveling Class Mark (TCM) used at ETN switches. (See Technical Publication 42709, "Tie Trunk Signaling Compatibility for Connecting to a DIMENSION PBX," July 1979.)

When a seven-digit/ten-digit UDP call or a DSC (Distant Steering Code) (CDP) call is made from a Meridian 1 Node to an ETN switch, the dialed digits together with the TCOS number (0-7) are sent to the connected ETN switch. At the ETN switch, the TCOS number received from the Meridian 1 Node is used as a TCM to determine route access and Off Hook Queuing eligibility at the ETN switch.

Similarly, when a call is made from an ETN switch to a Meridian 1 Node, the dialed digits together with the TCM number (0-7) are sent to the connected Meridian 1 Node. The Meridian 1 Node interprets the received TCM number as a TCOS number. The received TCM (for example, TCOS) replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch. This new FRL (for example, TCM) is then used to determine route access and Off Hook Queuing eligibility for the call. However, if a DSC (CDP) call is terminated on an SL-1 switch as a Local Steering Code (LSC) call, the TCOS value transmitted by the connected switch will not be collected and saved by this switch.

Facility Restriction Level

A Facility Restriction Level (FRL) number is associated with each NCOS group. The FRL ranges from 0 (low-privilege) to 7 (high-privilege). The FRL is used by the software to determine the Alternate Route Selection choices available for network call attempts by users within an NCOS group.

A user assigned to an NCOS with an FRL of 3 is allowed access to Alternate Route Selection choices that have an assigned FRL of 3 or less. Access to route choices with an FRL greater than 3 is denied. A user whose NCOS has an FRL of 7 has access to any Alternate Route Selection. By assigning low-privilege users to an NCOS with a low FRL, and high-privilege users to an NCOS with a higher FRL, the customer controls worker and management access to all network facilities.

Automatic On-Net to Off-Net Overflow

If all on-net facilities to a location are busy or blocked, NARS can convert a dialed NARS UDP number to the Listed Directory Number (LDN) or DID number of the destination location, and use off-net facilities to complete the call. If the Meridian 1 Node is equipped with BARS, digit manipulation tables can be used to convert the dialed number to the off-net number (LDN or DID) of the destination location.

A user at customer location I (Figure 2) dials 8-777-3283 to reach a party with extension number 3283 at customer location H. At the Node, NARS directs the dialed LOC number (777) to a route list, and searches all eligible routes in the list. Failing to find an available tie trunk route, NARS then seizes local off-net facilities and, to complete the call, outpulses one of two possibilities:

- 224-3283, if customer location H is arranged for DID
- 224-5600, if customer location H is not arranged for DID

Multiple DID Office Code Screening

Prior to X11 release 5, only one contiguous DID DN range can be defined per location. DNs outside the range are converted to the LDN.

With X11 release 5 and later, Multiple DID Office Code Screening enhances On-Net to Off-Net Overflow. This enhancement permits on-net calls that are routed through the public network using on-net to off-net conversion to terminate at any DN that has been defined in the Location Code memory data block. For each LOC defined, Multiple DID Office Code Screening allows the following:

- define multiple NXX codes
- define multiple ranges of DN within each NXX
- route calls to DID ranges with an NXX different from the LDN

The following arrangements of multiple office codes (NXX) and multiple DN ranges are possible:

- single office code with a single DN range
- single office code with multiple DN ranges
- multiple office codes with a single DN range
- multiple office codes with multiple DN ranges

Only one NPA per LOC code is allowed.

Ranges defined within an LOC code must be unique. Overlapping or duplication of ranges is not permitted.

The number of digits in each DID range must be four.

A maximum of 20 DID ranges may be defined per Location Code regardless of the number of office codes.

Incoming Trunk Group Exclusion

With X11 release 5 and later, Incoming Trunk Group Exclusion stops users on Meridian 1 and Conventional Mains from utilizing the network to reach destinations in the home NPA, or other restricted NPAs, NXXs, LOCs, and SPNs. When the feature is configured, users cannot use the network to circumvent the restrictions. Instead, they are forced to dial off-net from their own switch and become subject to whatever restrictions are imposed at the main.

Standard call blocking is applied on outgoing calls to specific NPAs, NXXs, SPNs or LOCs at the Meridian 1 Node if the call is from a specific incoming trunk group. Two advantages result:

- Loopback routing through the caller's home switch (home NPA, NXX) is prevented. Calls that should have been made off-net from the caller's home switch are blocked outgoing at the node.
- Main users are prevented from using BARS or NARS to make calls to certain NPA, NXX, SPN, or LOC that they are restricted from making at the home switch.

There is one Incoming Trunk Group Exclusion index (255 groups) for each defined NPA, NXX, SPN, or LOC. Each index points to an Incoming Trunk Group Exclusion (ITGE) table. A maximum of 128 restricted routes can be defined in each ITGE table. Incoming Trunk Group Exclusion provides 10-digit restriction for NPA codes, 11-digit restriction for SPN codes, 7-digit restriction for NXX codes, and 3-digit restriction for LOC codes. For LOC codes, the entire code can be restricted as well.

When a call is received, BARS or NARS tests to see if the dialed code is a restricted type (Supplemental Digit Restriction). If it is, BARS or NARS checks whether or not it has an ITGE restriction and if there is an index number (ITEI) associated with it. If an ITEI is defined, the ITGE table corresponding to the dialed code is searched. If the incoming trunk route is a member of the ITGE, the BARS or NARS process is terminated and the call is blocked. If Incoming Trunk Exclusion Digits (ITED) are assigned, BARS or NARS checks the ITED for a match with the dialed digits following the NPA, NXX, or SPN. If they match and the ITGE includes the originating trunk route, the call is blocked.

Off-Net Number Recognition

With X11 release 5 and later, Off-Net Number Recognition eliminates the need for using two extra CO trunks when a subscriber using the private network dials a DID or DDD number that terminates at a BARS or NARS location. Calls are routed directly to the dialed DN (DID calls) or to the LDN (DDD calls), rather than being switched from the terminating switch to the CO and back again.

Off-Net Number Recognition parameters for local and remote DDD and DID locations are defined by the customer in the Network Translation Tables, Supplemental Digit Restriction/Recognition blocks (SDRR), digit manipulation tables, and the Route Data Block. (See Tables 7 and 8.)

Up to ten digits can be defined for recognition with X11 releases 5, 6, and 7. X11 release 8 and later supports up to 11 digits.

Table 7
SDRR (X11 release 5, 6, and 7)

Call type	Network Translation Table (number of digits)	SDRR block (number of digits)
NPA	3	1–7
1NPA	4	1–7
NXX	3	1–4
1NXX	4	1–4
SPN	4	1–6
ISPN	4	1–7

Table 8
SDRR (X11 release 8 and later)

Call type	Network Translation Table (number of digits)	SDRR block (number of digits)
NPA	3–10	1–(10–N)
1NPA	4–11	1–(11–N)
NXX	3–7	1–(7–N)
1NXX	4–8	1–(8–N)
SPN	1–11	1–(10–N)
ISPN	1–4	1–(11–N)

Note: The value N is equal to the number of digits defined in the Network Translation Table.

Up to 512 SDRR blocks can be defined for NARS (256 for BARS). Each table can contain up to 64 entries.

Off-net numbers are recognized at the last intelligent BARS or NARS switch. Translating the NPA, NXX, or SPN identifies the method of call treatment. If the data type is SDRR and the index is an SDRR table index, supplemental digit restriction/recognition is applied:

- If no match is found in the SDRR, route selection is called, call processing resumes, and the call is routed to the CO of the terminating off-net number.
- If a match is found **and** the number is in the denied block, standard call blocking takes place.
- If a match is found **and** the number is recognized as a terminating number at the local switch (for example, the last intelligent BARS or NARS switch), the call is terminated at the station DN (DID calls) or at the attendant DN (DDD calls).
- If a match is found, **and** the dialed number is a recognized number terminating at a remote switch, **and** the switch has chosen a tie route for call termination, the switch checks the Route Data Block for special digit manipulation. The call is routed directly to the main. DID calls terminate at the dialed station, and DDD calls terminate at the attendant DN.

Note: Remote recognition applies to tie trunks only.

Supplemental Digit Restriction (release dependent)

Supplemental Digit Restriction blocks (see Table 9) function as follows:

– block (deny) access to certain telephone numbers

– recognize off-net calls dialed to on-net locations

– prevent routing of calls to the home switch of the originating trunk group by either on-net or off-net facilities

The customer can also specify through Meridian 1 service change the treatment that blocked calls receive, such as Overflow Tone, Intercept to Attendant, or Recorded Announcement.

Table 9
Supplemental Digit Restriction blocks per NARS or BARS

Network package	X11 release 4 and earlier maximum	X11 release 5 and later maximum
NARS	256	512
BARS	32	256

Note: For X11 release 4 and earlier, one block can restrict up to 16 numbers. For X11 release 5 and later, one block can recognize or restrict up to 64 numbers.

Digit translation/restriction/recognition

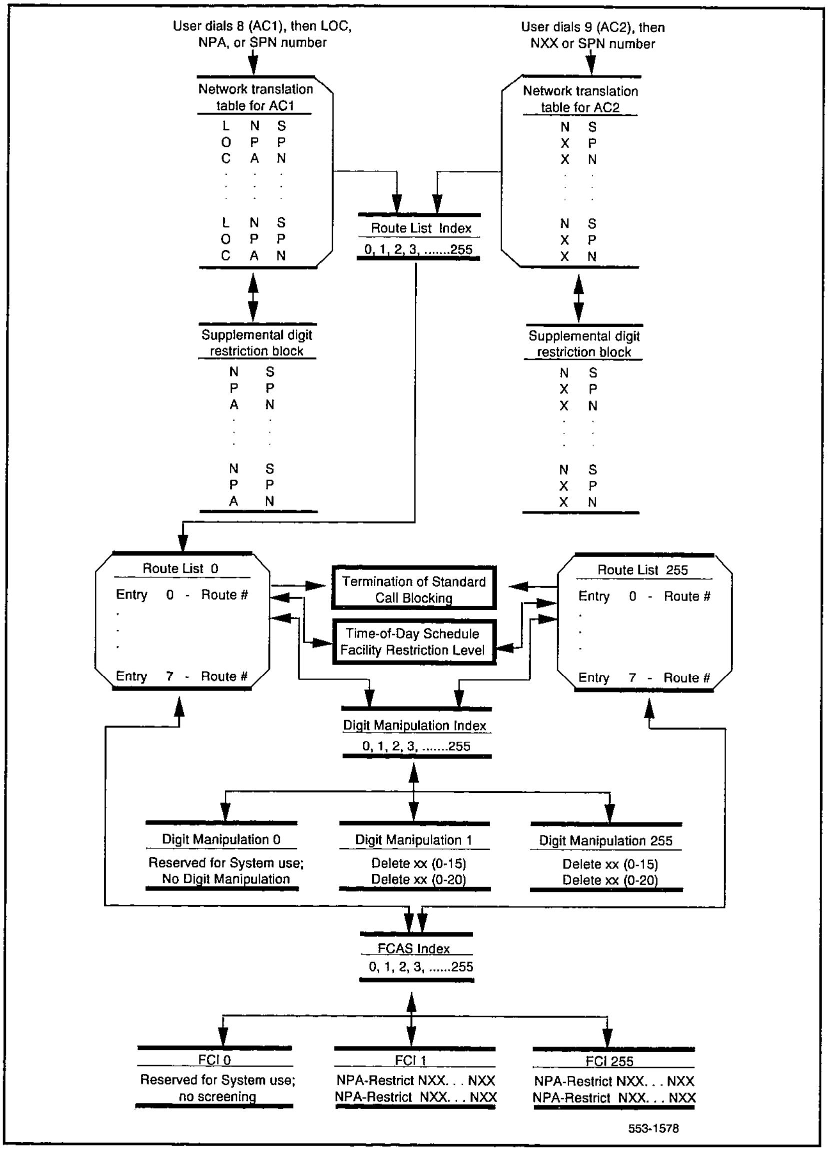
BARS and NARS provide a one- through ten-digit translation/restriction/ recognition capability through the use of Network Translation Tables and Supplemental Restriction/Recognition tables. There are two Network Translation Tables with NARS, one associated with each of the network access codes (AC1 and AC2). See Figure 3 for NARS elements accessed at a Meridian 1 node to process a network call.

Information contained in the Network Translation Tables is as follows:

- For each NPA entry, excluding the Home NPA (HNPA):
  - a route list index number (0-255 with NARS, 0-127 with BARS) that indicates which route list to use in processing a call to this NPA entry
  - whether there are telephone numbers within this NPA entry to which network calls are to be blocked (for example, denied)
  - whether there are telephone numbers within this NPA entry to which network calls are to be blocked because of ITGE restrictions (X11 release 5 and later)
  - whether there are numbers under this NPA entry that are to be recognized as DID or DDD codes to an on-net location (X11 release 5 and later)
  - a list (up to 64) of one- to seven-digit numbers that follow the NPA and are to be blocked or recognized in this NPA
- For each NXX entry:
  - a route list index number (0-255 with NARS; 0-127 with BARS) that indicates which route list to access in processing a call to this NXX entry
  - whether there are telephone numbers within this NXX entry to which network calls are to be blocked
  - whether there are telephone numbers within this NXX entry to which network calls are to be blocked because of ITGE restrictions (X11 release 5 and later)
  - whether there are numbers under this NXX entry that are to be recognized as DID or DDD codes to an on-net location (X11 release 5 and later)
  - a list of up to 64, one- to four-digit numbers that follow the NXX and are to be blocked or recognized in this NXX

- For each LOC entry, excluding the Home LOC code (not applicable to a BARS-equipped switch):
  - a route list index number (0-255) that indicates which route list to access in processing a call to this LOC entry
  - the Listed Directory Number (LDN) to which the LOC entry is to be converted when using off-net DDD facilities
  - the range of DID numbers to which the LOC entry can be converted when using DID facilities
  - office codes associated with the range
  - whether there are LOC entries to which network calls are to be blocked because of ITGE restrictions (X11 release 5 and later)
- For each SPN (special number) entry:
  - a route list index number (0-255 with NARS; 0-127 with BARS) that indicates which route list to access in processing a call to this SPN
  - whether there are digits following SPN numbers to which network calls are to be blocked
  - whether there are SPN numbers for blocking network calls because of ITGE restrictions (X11 release 5 and later)
  - whether there are numbers under this SPN that are to be recognized as DID or DDD codes to an on-net location (X11 release 5 and later)
  - a list of up to 64, one- to ten-digit numbers that are to be blocked or recognized when following the SPN

Figure 3
NARS elements accessed at a Meridian 1 node to process a network call



Free Calling Area Screening

Free Calling Area Screening (FCAS) is a BARS or NARS feature that provides full six-digit (NPA-NXX) screening to determine the route choice for completion of off-net calls. With FCAS, a customer can allow calls to NXX codes within the "free calling area" surrounding a particular on-net location and restrict (deny) calls to those NXX codes that would incur long distance charges.

FCAS is implemented similarly to digit manipulation (through tables). A Meridian 1 Node equipped with NARS can accommodate up to 255 FCAS tables; with BARS, up to 127 FCAS tables. Each table can contain up to 15 NPA codes. (Prior to X11 release 19, with BARS, each table can contain up to 7 NPA codes.)

Up to 800 NXX codes can be restricted or allowed within each NPA code. Each FCAS table is referenced by a Free Calling Index (FCI) number (0-255 with NARS, 0-127 with BARS); "FCI = 0" is a system default indicating that no Free Calling Area Screening is required. The appropriate FCI number is then assigned to the applicable route list entries.

Whenever a route list entry is being evaluated for an off-net call (for example, 8-NPA-NXX-XXXX), BARS or NARS checks to see if there is an FCI number (other than "0") referred to by the entry. If an FCI number other than "0" is defined, the appropriate FCAS table for the dialed NPA is found and used for NXX screening. If the dialed NXX is denied in the table, BARS or NARS will not use the route list entry for call completion, but will continue to search for another eligible route list entry. If the dialed NXX is not denied or specifically allowed in the table, the route list entry is eligible for the call. Calls to the LDN of a location are screened only if the NPA is included as part of the LDN. NXXs allowed in an FCI table are the only ones allowed for that route list entry.

BARS or NARS bypass control

A customer can allow selected users to bypass the BARS or NARS feature for call completion between any two locations; for example, two locations that share a high community of interest. To do this, routes and trunks are set up between the two locations, and assigned an access code distinct from the AC1 and AC2 codes used to access BARS or NARS. The normal trunk controls, like Trunk Group Access Restriction (TGAR), Class of Service, and Code Restriction, are then used to enable access only to the selected users. All other users are denied access to the trunk group, and thus are forced to use BARS or NARS for all calls.

Network Speed Call

The Network Speed Call (NSC) feature enables a user at a Meridian 1 Node who is normally restricted from making certain types of BARS or NARS calls to make such a call if the destination is a company-approved number defined in a System Speed Call (SSC) list. This feature requires that the System Speed Call feature (see *X11 features and services* (553-3001-305)) be equipped, in addition to Network Speed Call. NSC can also be accessed by users at a Meridian 1 Main or Conventional Main, provided a BARS or NARS access code is used to initiate the call.

Access to the NSC feature is allowed after a NARS or BARS access code is dialed. Upon receipt of the BARS or NARS dial tone (optional), the user dials a Network Speed Call access code (one to three digits). The NSC access code must be unique from all LOC, NPA, and NXX codes and Special Numbers defined in the translator for the BARS or NARS access code.

The NSC access code is associated with a previously defined System Speed Call list (0-4095) through service change in the network translation load. If the SSC list has its length (size) changed, the list access code and list number must be deleted and reentered into the NARS translator. Associated with the SSC list is an NCOS number. The NCOS assigned to the SSC list is applied to the call only if the FRL (0-7) is greater than that associated with the call originator's assigned NCOS.

> *Note:* With X11 release 13 and later, the number of SSC lists has expanded from 254 (0-253) to 4096 (0-4095), depending on available system memory and other speed call usage.

If 1+ dialing is specified for an NPA, NXX or SPN number in a translator, the digit 1 must not be used as the leading digit for Network Speed Call list codes in that translator.

The user then dials the number of the desired entry (0-999) in the SSC list. Upon completion of dialing, the digits defined for the list entry are passed to BARS or NARS translation for processing. Route and feature (OHQ, CBQ) eligibility for call completion are based on the NCOS associated with the SSC list, if the FRL of the user's assigned NCOS is lower than that of the list.

Network Call Transfer

This feature improves the operation of the existing Call Transfer (XFER) feature between two Meridian 1 systems when a call is transferred back to the originating switch. The regular XFER feature requires two tie trunks to complete the call. With Network Call Transfer (NXFER), if the call is transferred back to the originating switch by means of the same tie trunk group, the originating switch completes the transfer within itself and the tie trunks are dropped.

> *Note:* Both Meridian 1 switches must be equipped with NSIG and NXFER software for this feature to operate.

The benefits derived from the NXFER feature, which operates the same as the XFER feature, include:

- minimal use of access tie lines
- improved transmission performance because tie lines are not used for the completed connection

Figures 4 and 5 assume that station A receives an incoming trunk call from B and transfers it to C. As shown in Figure 4, the NXFER feature allows station A at one ESN switch (I) to transfer the tie trunk call from station B (switch II) to a third party, station C (switch II). In addition to NXFER software, NSIG software is needed at both ends of tie trunk. If the transfer is allowed, stations B and C are connected on switch II and the tie trunks are dropped (see Figure 6). In comparison, regular Call Transfer (XFER) requires two tie trunks and both switches to transfer and connect stations B and C.

Figure 4
Connection without NXFER

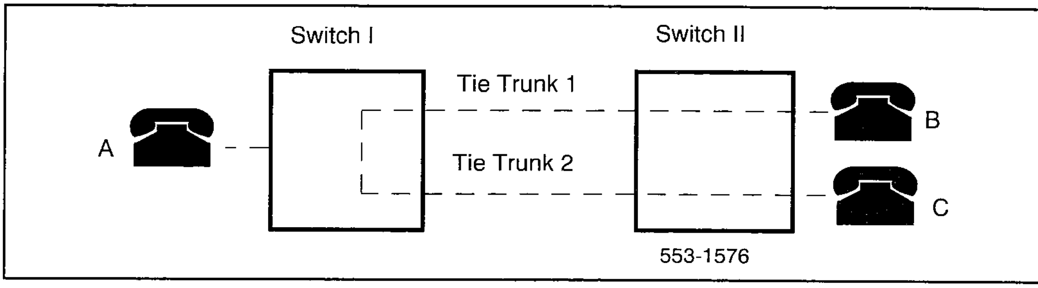


Figure 5
Connection during NXFER

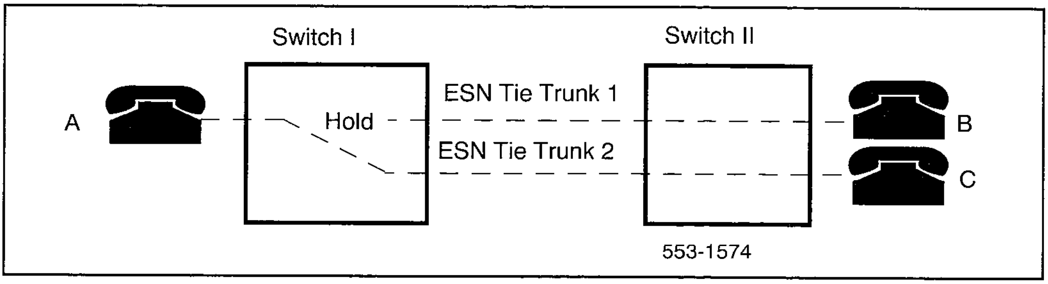
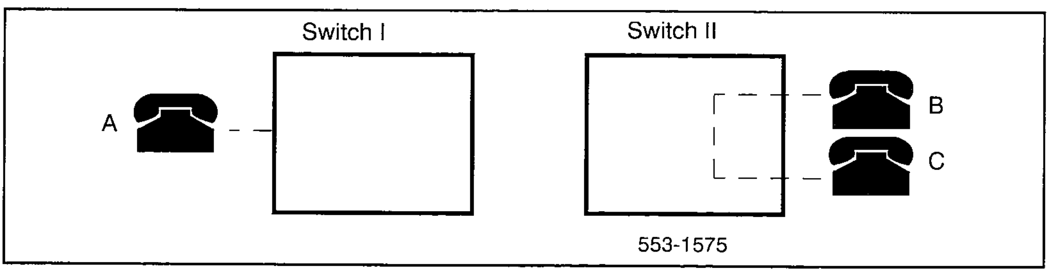


Figure 6
Connection after XFER



1+ dialing

Translation tables: With 1+ dialing, the BARS or NARS translation tables are equipped for 4-digit translation (based on the first one to four digits), or 11-digit translation with X11 release 8 on, thus allowing BARS or NARS access codes for long distance calls. NARS provides two translation tables and BARS provides one. Refer to Table 2 for dialing formats for NARS Uniform Dialing Plan (UDP) calls and Table 3 for BARS calls.

Code ambiguity: The 1+ dialing capability also eliminates ambiguity between identical originating and terminating three-digit NPA, NXX, and LOC codes for BARS or NARS originated calls. Thus, the BARS or NARS customer can route calls to any NPA, NXX, or LOC code that conflicts with one of his three-digit codes.

Limitation: If 1+ dialing is specified for an NPA, NXX, or SPN number in a translator, it cannot conflict with any NPA, NXX, or SPN in the same table.

Network signaling

The Network Signaling (NSIG) feature provides the required signaling protocol to interface Meridian 1 Nodes with Meridian 1 ESN Mains, Meridian 1 Nodes with other Meridian 1 Nodes, Meridian 1 Nodes with Conventional Mains, and Meridian 1 Nodes with Electronic Tie Network (ETN) switches.

When equipped with NSIG, a Meridian 1 Conventional Main is enhanced and becomes an ESN Main. When callers at an ESN Main place calls through a node or nodes with NSIG, their NCOS or TCOS travel with the call and are interpreted at other Meridian 1 switches that are equipped with NSIG. The tie trunk settings determine and control the operation of this feature.

When the NSIG feature is equipped at a switch, options are available (Route Data Block, LD16) to define the signaling arrangements between that switch and any other switch that may be connected to it by means of tie trunks. These options define what call information is to be transmitted to a connected switch and what call information is to be received from a connected switch. The option selected depends on the type of connected switch (Node, Main, Conventional Main, ETN) and the options (for example, CCBQ, CBQCM) that are available to the connected switch.

The signaling options are STD (standard), ESN (Electronic Switched Network), ESN2, ESN3, ESN5, and ETN (Electronic Tie Network).

STD: Arranges the tie trunk group for transmission/reception of the called number between switches. Sends outpulsed digits.

ESN (X11 release 2 only): Arranges the tie trunk group for transmission/reception of the call type, NCOS/TCOS, and called number between switches.

ESN2 (X11 release 3 and later): Arranges the tie group as described for ESN in X11 release 2. Used unless switch has NXFER or Satellite Link Control (SAT).

ESN3 (X11 release 3 or 4): Arranges the tie group as described for ESN in X11 release 2; is required on systems equipped with the Network Call Transfer (NXFER) or Satellite Link Control features.

ESN5 (X11 release 5 and later): Arranges the tie trunk group as described for ESN in X11 release 2; needed with DTI.

ETN: Arranges the tie trunk group for transmission/reception of the called number and TCOS/TCM between switches and is used when connected to an ETN switch. Sends outpulsed digits and TCOS.

Application

Following is a description of how these options accommodate the different switch types that can be connected to a Meridian 1 Main or Meridian 1 Node that is equipped with the NSIG feature.

Meridian 1 Node: A Meridian 1 Node can be connected by means of tie trunks to another Meridian 1 Node, a Meridian 1 Main, a Conventional Main, and/or an ETN switch.

- If the Meridian 1 Node connects to another Meridian 1 Node, both ends of the connecting tie trunk group are defined with the ETN option (the called number plus TCOS/TCM), ESN2, ESN3, or ESN5.
- If the Meridian 1 Node connects to a Meridian 1 Main, both ends of the connecting tie trunk group are defined with the ESN option (call type plus NCOS/TCOS plus the called number).

- If the Meridian 1 Node connects to a Conventional Main, the Node-end of the tie trunk group is defined with the STD option (the called number).
- If the Meridian 1 Node connects to an ETN switch, the Node-end of the tie trunk group is defined with the ETN option (the called number plus TCOS TCM).

Meridian 1 ESN Main: A Meridian 1 Main can be connected by means of tie trunks to a Meridian 1 Node and satellite switches.

For connection to a Meridian 1 Node, both ends of the connecting tie trunk group are defined with the ESN option (call type plus NCOS/TCOS plus the called number).

- If there are satellite switches connected to the Meridian 1 Main, the main-end of the tie trunk groups from the satellite switches are defined with the STD option (the called number).

Requirements

The following requirements apply:

- As a Meridian 1 Main can connect to only one Meridian 1 Node, both switches must be equipped with the NSIG feature for NSIG-related features.
- Tie trunks between Meridian 1 Nodes and Meridian 1 Mains must be arranged for DTMF sending/receiving and wink-start operation.
- Meridian 1 Node compatibility with ETN switches is limited to seven-digit on-network, ten-digit off-network, and DSC (CDP) calls.

Satellite Link Control

Tandem trunk calls, when connected through more than one communications satellite trunk, are subject to transmission distortion because of propagation to and from communications satellites. The Satellite Link Control feature ensures that the configuration of a call does not include more than one communications satellite trunk.

When accessing an SCC, the SL-1TD is instructed to look for one of the following tones:

- Busy or Overflow Tone
- SCC dial tone or regular dial tone
- Ringback tone

This feature applies to ESN network calls (BARS or NARS/CDP) only.

ESN Proprietary Signaling (NSIG) is required among ESN switches.

Routes that receive digits from satellites or send digits to satellites have to be marked as SATELLITE routes for this feature to operate.

Simultaneously a timer is started whose value is an approximation of the period in which Ringback tone is normally received. When the timer expires prior to a response from the SL-1TD, the software sends a status request message to the SL-1TD. The response from the SL-1TD indicates that a tone has been detected but is not yet identified. Based on the sequence in which tones are checked, the software assumes the tone under investigation is Ringback tone and can continue the SCC call processing.

Flexible ESN "0" Routing

Flexible ESN "0" Routing, available beginning with X11 release 16, uses four prefixes to call the local operator (prefix of 0) or international operator (prefix 00), or to make station to station international calls (011), or calling card, collect, or other operator-assisted international calls (01).

Normally, the ESN translation table only contains leftwise unique numbers (for example, if one entry begins with the digits "123," no other entry can begin with the digits "123." These four special "0" prefixes, which are not leftwise unique, are an exception to this rule.

Flexible ESN "0" Routing is part of the existing BARS (57) and NARS (58) packages and interacts only with these features. Since NARS has two translation tables, two Flexible ESN "0" Routing data blocks are included in NARS. A call could be routed two different ways.

Flexible ESN "0" Routing applies to all route types and network types that are supported by ESN. For information on the appropriate prompts and responses in service change (LD90), refer to *X11 input/output guide* (553-3001-400).

Network traffic measurements

The Network Traffic (NTRF) feature provides Traffic Measurement data related to network performance and network traffic at each Meridian 1 Node and Meridian 1 Main. Effective use of this data enables the network communications manager to assess the effectiveness of the network, and to identify specific areas of network operation where improvements are warranted.

Areas of network operation that are measured include NCOS, OHQ, CBQ, and routing. These measurements are provided in addition to existing non-network traffic data in *Traffic measurement formats and output* (553-2001-450).

TFN001 routing measurements

The routing measurements provide data related to route list utilization. A route list is a list of alternate trunk routes that are identified through interpretation of the dialed number. For each defined route list, these measurements show:

- how often the list was used
- which routes in the list were used
- the number of calls that were unsuccessful in completing a route list selection or connection
- queuing (OHQ and CBQ) information

These data comprise both the usage and average duration of each call against the route list. Data is output only for route lists upon which calls were attempted during the traffic study interval.

OHQ measurements

The OHQ measurements are associated with route lists, NCOS, and incoming trunk groups. OHQ enables a user to wait off hook for a network facility to become available. The network communications manager can control the conditions for, and the duration of, the wait. The measurements present both the usage and average duration if the OHQ feature. This data comprises part of the routing measurements.

CBQ measurements

The CBQ measurements are associated with route lists, NCOS, and incoming trunk groups. If the system cannot select a route, the caller may opt to have the system establish the call at a later time when a route becomes available. The measurements indicate usage and the length of time a caller waits for an established call. The communications manager can control the conditions for CBQ and the advancement of a call within the queue. This data comprises part of the routing measurements.

TFN002 NCOS measurements

The Network Class of Service (assigned through service change) indicates the network facilities available to the user. Traffic Measurements are accumulated for each NCOS and indicate the grade of service (queuing and blocking delay, for example) provided by the system. The communications manager can change the NCOS if a grade of service is inappropriate for a user category, the definition of the NCOS is inaccurate, or the routing parameters need updating.

TFN003 Incoming Trunk Group Measurements

These measurements relate to the incremental traffic that was imposed on the incoming trunk groups by network queuing features (for example, OHQ, CBQ, CCBQ, CBQCM).

The measurements show:

- how often the incoming trunk group was offered OHQ
- the average time in the OHQ
- how often the incoming trunk group was offered CCBQ or CBQCM, and the number of acceptances
- the average time in CCBQ or CBQCM
- how often an access line was blocked when attempting a CCBQ or CBQCM call back
- how often CCBQ or CBQCM call backs were not answered or were canceled

Expensive Route Warning Tone

In some cases, expensive trunk routes may be assigned an FRL that allows the routes to be accessed by certain network users. This feature enables the network communications manager to identify users who should receive an Expensive Route Warning Tone (ERWT). The tone (three 256-ms bursts of 440 Hz) notifies the user that BARS or NARS has selected facilities designated as expensive to complete the call. The user then has the choice of allowing the call to complete over the expensive facilities, going on hook to avoid the increased expense, or queuing on the I set routes. The user must make this choice within a programmable time of 0 to 10 seconds.

If the call originator is located at a Meridian 1 Node or Meridian 1 Main and the user is eligible for both the Ring Again feature and for CBQ type A, then Ring Again may be activated to queue the call. See *Network Queue description* (553-2751-101).

If the Meridian 1 Node is equipped for Call Detail Recording (CDR), acceptance of an expensive route after ERWT is received is noted in the CDR record.

ERWT eligibility is defined through the user's assigned NCOS group. NCOS groups must have CBQ type A in order to hear ERWT. The tones must be activated for the customer group. The expensive route cannot be part of the I set in the route list. Refer to the *X11 input/output guide* (553-3001-400), LD86, and the ERT timer for more information.

Queuing features

Eligibility for the Off Hook Queuing (OHQ) or Call Back Queuing (CBQ) features is defined through the user's assigned NCOS group. See *Network Queue description* (553-2751-101).

BARS implementation

This section provides the procedures necessary to configure Basic Alternate Route Selection (BARS). Only the service change information for BARS is shown here. For a complete discussion of prompts and responses, see *X11 input/output guide* (553-3001-400). The following procedure shows the steps that must be performed to correctly implement your BARS system.

Procedure 1
Implementing BARS

1. Gather data for each NCOS group (LD87).
2. Gather data to define BARS feature parameters (LD86).
3. Gather data for each Digit Manipulation Index (LD86).
4. Gather data for each Free Calling Area Screening table (LD87).
5. Gather data for each route list associated with a Digit Manipulation table (LD86).
6. Gather data for each Incoming Trunk Group Exclusion list (LD86).
7. Gather BARS translation data (LD90).
8. Gather data to configure a Conventional Main for Off-Net Number Recognition (LD16).
9. Gather data to assign a Network Class of Service group number to a list of items (LDs 10, 11, 12, 14, 24, 18, 88).
10. Enter data into the database.

Configuring BARS

Step 1
Gather data for each NCOS group. Access LD87.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NCTL	Network Control
NCOS	0-7 0-99	NCOS group to be added, changed, or removed
FRL	0-7	Facility Restriction Level
RWTA	Yes, (No)	Expensive Route Warning Tone

Step 2
Gather data to define BARS feature parameters. Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ESN	Electronic Switched Network data block
MXSD	0-512	Maximum number of Supplemental Digit Restriction blocks
MXIX	0-127	Maximum number of Incoming Trunk Group Exclusion tables that can be defined
MXDM	0-256	Maximum number of digit manipulation tables
MXRC	0-255	Maximum number of Free Calling Area Screening tables
MXRL	0-127	Maximum number of route lists
AC1	xx	One- or two-digit BARS access code one
DLTN	No, (Yes)	Dial tone after dialing AC1 or AC2
ERWT	No, (Yes)	Expensive Route Warning Tone is not (is) provided
ERDT	0-(6)-10	Time in seconds that a user has to accept or reject an expensive route after ERWT is given; default is 6 seconds; range is in 2-second intervals.
TODS	x hh mm hh mm	Start and end Time of Day schedules; BARS or NARS = 0-7 x = schedule number hh = hour mm = minutes
TGAR	Yes, (No)	Check for Trunk Group Access Restrictions Yes = examine TGAR/TARG when call is placed No = ignore TGAR/TARG when call is placed

Step 3
Gather data for each Digit Manipulation Index (DMI). Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	DMI	Digit Manipulation Index
DMI	1-255	Digit manipulation table index (see note below)
DEL	(0)-15	Number of leading digits to be deleted from the dialed number; default is 0.
INST	x...x	Up to 24 leading digits to be inserted

Note: The maximum number of digit manipulation tables is defined by prompt MXDM.

Step 4
Gather data for each Free Calling Area Screening (FCAS). Access LD87.

REQ	NEW, CHG, PRT	Action request
CUST	0-99	Customer number
FEAT	FCAS	Free Calling Area Screening
FCI	1-127	Free Calling Area Screening table index number
NPA	xxx	Three-digit NPA code to be screened. 7 NPA codes per table prior to X11 release 19; 15 with that and later releases.
NXX	DENY, ALOW	Allow or Deny NXX codes for NPA
DENY	xxx xxx	NXX code or range of codes to be denied
ALOW	xxxx xxxx	NXX code or range of codes to be allowed

Step 5
Gather data for each route list associated with A DMI. Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	RLB	Route list data block
RLI	0-127	Route list index number
ENTR	0-63	Route list entry number
ROUT	0-511	Route number associated with the index
TDET	Yes, (No)	Tone detector is (is not) used
TYPE	TIE, (CC1), CC2	TIE = Tone detector for on-net calls (CC1) = Tone detector for special common carrier (SCC), Type 1 CC2 = Tone detector for special common carrier (SCC), Type 2
TONE	DIAL, (SCC)	Type of tone expected from SCC DIAL = normal dial tone SCC = SCC dial tone
TOD	0-7 X0-7	Time of Day schedule associated with the entry Turn off a time of day schedule
EXP	Yes, (No)	Entry is (is not) classed as expensive
FRL	(0)-7	Minimum Facility Restriction Level a user must have to access the entry
DMI	(0)-255	Index number of the digit manipulation table to be used for the entry; default is 0, no digit manipulation required.
FCI	1-127	Free Calling Area Screening table index number (FCAS) (0) = no FCAS required
OHQ	Yes, (No)	Off Hook Queuing is (is not) allowed on the entry
CBQ	Yes, (No)	Call Back Queuing is (is not) allowed on the entry
ISET	(0)-32	Number of entries in the initial route set
MFRL	(MIN), 0-7	Minimum FRL used to determine autocode prompting

Step 6
Gather data for each Incoming Trunk Group Exclusion index (ITGE). Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ITGE	Incoming Trunk Group Exclusion data block
ITEI	1-127	Incoming Trunk Group Exclusion index number; if REQ was "OUT" all route entries defined for the entered index are removed.
	<return>	Return to REQ prompt
RTNO	0-511	Route number associated with index; precede with an X to delete an existing route.
	<return>	Return to REQ prompt

Step 7
Gather BARS translation data. Access LD90. (Part 1 of 4)

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NET	Network Translation Tables
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables
TYPE	LOC HNPA NPA HLOC NXX SPN NSCL	Location Code Home NPA translation code Numbering Plan Area translation code Home Location Code Central office translation code Special Number translation code Speed Call
	<return>	Return to REQ

The following prompt occurs if TYPE is LOC:

LOC	x...x	Location Code (3 digits) or extended LOC (3 to 7)
RLI	0-127	Route list index
ITEI	(0)-255	Incoming Trunk Group Exclusion index
LDN	xx...xx	Up to ten-digit listed Directory Number including NPA
DID	Yes, (No)	This location arranged for DID
MNXX	Yes, (No)	Multiple NXX codes and ranges
SAVE	1-4	Number of trailing digits to be saved in dialed extension number; must be four if MNXX = Yes.
OFFC	xxx	NXX of the DID number; prompted if MNXX = Yes.
RNGE	0-9999 0-9999	Bottom and top of range of DID numbers

Step 7
Gather BARS translation data. Access LD90. (Part 2 of 4)

The following prompts occur if TYPE is NPA:

NPA	xxx, xxx yyy	Area code translation, extended NPA code translation
RLI	0-127	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN. The maximum number of digits allowed is 7-m.
ITEI	x...x	Incoming Trunk Group Exclusion index

Legend:
m = the number of digits entered for the prompt NPA
X = a command used to clear a recognized xxx code
<return> = A return after each subprompt takes you back to SDRR

Step 7
Gather BARS translation data. Access LD90. (Part 3 of 4)

The following prompts appear if TYPE is NXX:

NXX	xxx, xxx yyy	Office code translation, extended NXX codes translation
RLI	0-127	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR. The maximum number of digits allowed is 7-m.
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN
ITEI	(0)-255	Incoming Trunk Group Exclusion index

The following prompts appear if TYPE is SPN:

SPN	xxxx xxxx x..	Special Number translation
RLI	0-127	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR
DMI	1-255	Digit manipulation table index
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN
ITEI	(0)-255	Incoming Trunk Group Exclusion index

Step 7
Gather BARS translation data. Access LD90. (Part 4 of 4)

DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN
ITEI	(0)-255	Incoming Trunk Group Exclusion index

The following prompts appear if TYPE is NSCL:

NSCC	xxx	One- to three-digit Network Speed Call access code
SSCL	0-4095	System Speed Call list number

Step 8
Gather data to configure a Conventional Main for Off-Net Number Recognition. Access LD16.

REQ	NEW, CHG	Add or change a route
TYPE	RDB	Route data block
CUST	0-99	Customer number
ROUTE	0-511	Route number
TKTP	TIE	Tie trunk
CNVT	Yes, (No)	Route to Conventional switch (prompted if the response to TKTP is Tie)
DDMI	(0)-127	Digit Manipulation Index (prompted if the response to CNVT is Yes)
ATDN	xxxx	Attendant DN of Conventional Main (prompted if the response to CVNT is Yes)

Note: If the DN expansion package is equipped, the attendant DN can have up to seven digits.; otherwise, only four digits can be entered.

The following procedure involves several overlay programs. In all cases, the prompt is NCOS, the range is 0-99 (0-7 prior to X11 release 13), and the default is 0.

Step 9
Gather data to assign a Network Class of Service group number to a list of items.

- 500/2500 telephones (LD10)
- Meridian Modular telephones, SL-1 telephones, M1000 series telephones, M2000 series digital telephones, M3000 Touchphones (LD11)
- Attendant consoles (LD12)
- Trunks (LD14)
- Direct Inward System Access Directory Number (LD24)
- System Speed Call (LD18)
- Authorization Code (LD88)

Step 10
Enter data into the database.
The final step in configuring Basic Alternate Route Selection is to enter the data gathered into the database.

1 Log in.

2 Load the appropriate program.

3 Enter data requested by prompts until REQ prompt returns.

4 More data to be added/removed?

Yes = go to Step 2.

No = perform datadump.

Note: A datadump takes approximately 3 minutes to complete. If large amounts of data are being added, perform a datadump periodically. This ensures that entered data is not lost should a system reload occur.

5 Print data (respond with PRT to prompt REQ) and verify against data forms.

6 Corrections required?

Yes = go to Step 2.

No = perform applicable tests.

NARS implementation

This section provides the procedures necessary to configure Network Alternate Route Selection.(NARS). Only the service change information for NARS is shown here. For a complete discussion of prompts and responses, see *X11 input/output guide* (553-3001-400). The following procedure shows the steps that must be performed to correctly implement your NARS system.

Procedure 2
Implementing NARS

1. Gather data for each NCOS group (LD87).
2. Gather data to define NARS feature parameters (LD86).
3. Gather data for each Digit Manipulation Index (LD86).
4. Gather data for each Free Calling Area Screening table (LD87).
5. Gather data for each route list associated with a Digit Manipulation table (LD86).
6. Gather data for each Incoming Trunk Group Exclusion list (LD86).
7. Gather NARS translation data (LD90).
8. Gather data to configure a Conventional Main for Off-Net Number Recognition (LD16).
9. Gather data to assign a Network Class of Service group number to a list of items (LDs 10, 11, 12, 14, 24, 18, 88).
10. Enter data into the database.

Configuring NARS

Step 1
Gather data for each NCOS group. Access LD87.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NCTL	Network Control
NCOS	0-99	NCOS group to be added, changed, or removed
FRL	(0)-7	Facility Restriction Level
RWTA	Yes, (No)	Expensive Route Warning Tone

Step 2
Gather data to define NARS feature parameters. Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ESN	Electronic Switched Network data block
MXLC	0-999	Maximum number of Supplemental Digit Restriction blocks
MXSD	0-512	Maximum number of Incoming Trunk Group Exclusion tables that can be defined
MXIX	0-255	Maximum number of Incoming Trunk Group Exclusion tables
MXDM	0-256	Maximum number of digit manipulation tables
MXRL	0-256	Maximum number of route lists
MXFC	0-255	Maximum number of Free Calling Area Screening tables
AC1	xx	One- or two-digit NARS/BARS access code one
AC2	xx	One- or two-digit NARS access code two
DLTN	No, (Yes)	NARS/BARS dial tone after dialing AC1 or AC2
ERWT	No, (Yes)	Expensive Route Warning Tone is not (is) provided
ERDT	0-(6)-10	Time in seconds that a user has to accept or reject an expensive route after ERWT is given; default is 6 seconds; range is in 2-second intervals.
TODS	x hh mm hh mm	Start and end Time of Day schedules; BARS or NARS = 0-7 x = schedule number hh = hour mm = minutes
TGAR	Yes, (No)	Check for Trunk Group Access Restrictions Yes = examine TGAR/TARG when call is placed No = ignore TGAR/TARG when call is placed

Step 3
Gather data for each Digit Manipulation Index (DGT). Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	DGT	Digit Manipulation Index
DMI	1-255	Digit manipulation table index (see note below)
DEL	(0)-15	Number of leading digits to be deleted from the dialed number; default is 0.
INST	x...x	Up to 24 leading digits to be inserted

Note: The maximum number of digit manipulation tables is defined by prompt MXDM.

Step 4
Gather data for each Free Calling Area Screening (FCAS). Access LD87.

REQ	NEW, CHG, PRT	Action request
CUST	0-99	Customer number
FEAT	FCAS	Free Calling Area Screening
FCI	1-255	Free Calling Area Screening table index number
NPA	xxx	Three-digit NPA code to be screened. 15 NPA codes supported per table.
NXX	DENY, ALOW	Allow or Deny NXX codes for NPA
DENY	xxx xxx	NXX code or range of codes to be denied
ALOW	xxxx xxxx	NXX code or range of codes to be allowed

Step 5
Gather data for each route list associated with A DGT. Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	RLB	Route list data block
RLI	0-255	Route list index number
ENTR	0-63	Route list entry number (X11 release 15 and later)
ROUT	0-511	Route number associated with the index
TDET	Yes, (No)	Tone detector is (is not) used
TYPE	TIE, (CC1), CC2	TIE = Tone detector for on-net calls (CC1) = Tone detector for special common carrier (SCC), Type 1 CC2 = Tone detector for special common carrier (SCC), Type 2
TONE	DIAL, (SCC)	Type of tone expected from SCC DIAL = normal dial tone SCC = SCC dial tone
TOD	0-7 X0-7	Time of Day schedule associated with the entry Turn off a Time of Day schedule
EXP	Yes, (No)	Entry is (is not) classed as expensive
FRL	(0)-7	Minimum Facility Restriction Level a user must have to access the entry
DMI	(0)-255	Index number of the digit manipulation table to be used for the entry; default is 0, no digit manipulation required.
FCI	(0)-255	Free Calling Area Screening table index number (FCAS) (0) = no FCAS required
OHQ	Yes, (No)	Off Hook Queuing is (is not) allowed on the entry
CBQ	Yes, (No)	Call Back Queuing is (is not) allowed on the entry
ISET	(0)-64	Number of entries in the initial route set
MFRL	(MIN), 0-7	Minimum FRL used to determine autocode prompting

Step 6
Gather data for each Incoming Trunk Group Exclusion Index (ITGE). Access LD86.

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	ITGE	Incoming Trunk Group Exclusion data block
ITEI	1-127	Incoming Trunk Group Exclusion index number; if REQ was "OUT" all route entries defined for the entered index are removed.
RTNO	0-511	Route number associated with index; precede with an X to delete an existing route.

Step 7
Gather NARS translation data. Access LD90. (Part 1 of 4)

REQ	NEW, CHG, OUT	Create, change, or remove data
CUST	0-99	Customer number
FEAT	NET	Network Translation Tables
TRAN	AC1, AC2, SUM	Access code 1, 2, or summary tables
TYPE	LOC HNPA NPA HLOC NXX SPN NSCL	Location Code Home NPA translation code Numbering Plan Area translation code Home Location Code Central office translation code Special Number translation code Speed Call
	<return>	Return to REQ

The following prompt occurs if TYPE is LOC:

LOC	x...x	Location Code (3 digits) or extended LOC (3 to 7)
RLI	0-255	Route list index
ITEI	(0)-255	Incoming Trunk Group Exclusion index
LDN	xx...xx	Up to ten-digit listed Directory Number including NPA
DID	Yes, (No)	This location arranged for DID
MNXX	Yes, (No)	Multiple NXX codes and ranges
SAVE	1-4	Number of trailing digits to be saved in dialed extension number; must be four if MNXX = Yes.
OFFC	xxx	NXX of the DID number; prompted if MNXX = Yes.
RNGE	0-9999 0-9999	Bottom and top of range of DID numbers

Step 7
Gather NARS translation data. Access LD90. (Part 2 of 4)

The following prompt occurs if TYPE is HNPA:		
HNPA	xxx	Home NPA
The following prompts occur if TYPE is NPA:		
NPA	xxx, xxx yyy	Area code translation, extended NPA code translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN. The maximum number of digits allowed is 10-m (11-m for 1+ dialing).
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN. The maximum number of digits allowed is 7-m.
ITEI	(0)-255	Incoming Trunk Group Exclusion index

Legend:
m = the number of digits entered for the prompt NPA
X = a command used to clear a recognized xxx code
<return> = A return after each subprompt takes you back to SDRR

Step 7
Gather NARS translation data. Access LD90. (Part 3 of 4)

The following prompts appear if TYPE is NXX:

NXX	xxx, xxx yyy	Office code translation, extended NXX codes translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index
DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR. The maximum number of digits allowed is 7-m.
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN
ITEI	(0)-255	Incoming Trunk Group Exclusion index

The following prompts appear if TYPE is SPN:

SPN	xxxx xxxx x..	Special Number translation
RLI	0-255	Route list index
SDRR	aaa	Type of supplemental restriction or recognition
DMI	1-255	Digit manipulation table index

Step 7
Gather NARS translation data. Access LD90. (Part 4 of 4)

DENY	x...x	Number to be denied within the NPA, NXX, SPN, or SDRR
LDID	x...x	Local DID number recognized within the NPA, NXX, or SPN
LDDD	x...x	Local DDD number recognized within the NPA, NXX, or SPN
DID	x...x	Remote DID number recognized within the NPA, NXX, or SPN
DDD	x...x	Remote DDD number recognized within the NPA, NXX, or SPN
ITED	x...x	Incoming Trunk Group Exclusion codes for NPA, NXX, or SPN
ITEI	(0)-255	Incoming Trunk Group Exclusion index
The following prompts appear if TYPE is NSCL:		
NSCC	xxx	One- to three-digit Network Speed Call access code
SSCL	0-4095	System Speed Call list number

Step 8
Gather data to configure a Conventional Main for Off-Net Number Recognition. Access LD16.

REQ	NEW, CHG	Add or change a route
TYPE	RDB	Route data block
CUST	0-99	Customer number
ROUTE	0-511	Route number
TKTP	TIE	Tie trunk
CNVT	Yes, (No)	Route to Conventional switch (prompted if the response to TKTP is Tie)
DDMI	(0)-255	Digit Manipulation Index (prompted if the response to CNVT is Yes)
ATDN	xxxx	Attendant DN of Conventional Main (prompted if the response to CVNT is Yes)

Note: If the DN expansion package is equipped, the attendant DN can have up to seven digits.; otherwise, only four digits can be entered.

The following procedure involves several overlay programs. In all cases, the prompt is NCOS; the range is 0-99 (or 0-7 with X11 release 12 and earlier); the default is 0.

Step 9
Gather data to assign a Network Class of Service group number to a list of items.
Gather data to assign a Network Class of Service group number to each of the following items:

- 500/2500 telephones (LD10)
- Meridian Modular telephones, SL-1 telephones, M1000 series telephones, M2000 series digital telephones, M3000 Touchphones (LD11)
- Attendant consoles (LD12)
- Trunks (LD14)
- Direct Inward System Access Directory Number (LD24)
- System Speed Call (LD18)
- Authorization Code (LD88)

Step 10
Enter data into the database.
The final step in configuring Basic Alternate Route Selection is to enter the data gathered into the database:

1 Log in.

2 Load the appropriate program.

3 Enter data requested by prompts until REQ prompt returns.

4 More data to be added/removed?

Yes = go to Step 2.

No = perform datadump.

Note: A datadump takes approximately 3 minutes to complete. If large amounts of data are being added, perform a datadump periodically. This ensures that entered data is not lost should a system reload occur.

5 Print data (respond with PRT to prompt REQ) and verify against data forms.

6 Corrections required?

Yes = go to Step 2.

No = Refer to 553-2YY1-230 and perform applicable tests.

List of terms

This section provides an alphanumeric list of the mnemonics used in this document and their related terms.

BARS

Basic Alternate Route Selection

CBQ

Call Back Queuing

CDP

Coordinated Dialing Plan

CDR

Call Detail Recording

DDD

Direct Distance Dialing

DMI

Digit Manipulation Index

DID

Direct Inward Dialing

ERWT

Expensive Route Warning Tone

ESN

Electronic Switched Network

ETN

Electronic Tie Network

FCAS

Free Calling Area Screening

FRL

Facility Restriction Level

ITGE

Incoming Trunk Group Exclusion

LDN

Listed Directory Number

NARS

Network Alternate Route Selection

NCOS

Network Class of Service

NFCR

New Flexible Code Restriction

NSC

Network Speed Call

NSIG

Network Signaling

NXFER

Network Call Transfer

OHQ

Off Hook Queue

RLI

Route List Index

SDR

Supplemental Digital Restriction

SDRR

Supplemental Digital Restriction/Recognition

SIGO

Signaling Arrangement

SPN

Special Number

TCOS

Traveling Class of Service

TGAR

Trunk Group Access Restriction

TOD

Time of Day

TTTN

tandem tie trunk network

UDP

Uniform Dialing Plan

XFER

Call Transfer

SL-1

Basic and Network Alternate Route Selection

Description

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SL-1

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Contents

About this document

This Northern Telecom Publication describes the queuing features and options that are available to switches which operate in a private network environment. The queuing features are:

- Off-Hook Queuing (OHQ)
- Call-Back Queuing (CBQ)
- Coordinated Call-Back Queuing (CCBQ)
- Call-Back Queuing to Conventional Mains (CBQCM)

Customer benefits from the queuing features include increased trunk group efficiency, facility cost-savings and user convenience through elimination of numerous redialing attempts.

A private network consists of one or more of the following types of switch.

Meridian 1 Node

A Meridian 1 switch that is equipped with either the Basic Alternate Route Selection (BARS) or Network Alternate Route Selection (NARS) software feature packages. NARS and BARS features are described in *Basic and network alternate route selection description* (553-2751-100).

Meridian 1 Main

A Meridian 1 switch that is connected via tie trunks to a Meridian 1 Node and is equipped with the Network Signaling (NSIG) and Network Class of Service (NCOS) software packages. A Meridian 1 Main can be connected to only one Meridian 1 Node. A Meridian 1 Main may also be equipped with the BARS and OHQ features.

Conventional Main

A switch (Meridian 1 or any other type) that is connected via tie trunks to a Meridian 1 Node and is not equipped with any network-related features.

> *Note:* Information presented in this practice assumes the reader is familiar with the terminology and concepts of the alternate route selection capabilities as described in *Basic and network alternate route selection description* (553-2751-100).

Off-Hook Queuing

Off-Hook Queuing (OHQ) is a software feature that can be equipped at a Meridian 1 Node and/or Meridian 1 Main. The feature enables a call originator to remain off-hook for a short time (customer programmable) until a network facility for call completion becomes available at the Node or Main.

OHQ eligibility

Network calls may be placed in an OHQ if all trunk routes (entries) in the initial route set of a route list (see Note) are busy and the following criteria are met.

- At least one of the trunk routes in the initial route set of a route list is defined as being eligible for OHQ.
- The Network Class of Service (NCOS) of the call originator (at a Node or a Main) is defined to permit OHQ.
- The incoming trunk group at the Node or the Main is defined in software to permit OHQ for incoming calls.
- The Traveling Class Mark (TCM) received at the Node from an Electronic Tie Network (ETN) switch is compatible with a Facility Restriction Level (FRL) that is defined to permit OHQ.
- The Traveling Class of Service (TCOS) received at the Node from another Node is compatible with an FRL that is defined to permit OHQ.
- The probability that the call will timeout in the OHQ before a trunk becomes available is below a specific threshold.
- The OHQ feature is enabled for the customer.

Note: A Meridian 1 defined initial set "marker" determines which route list entries are in the initial route set. Typically, the initial route set contains the cheaper routes to a particular destination. The remaining routes in the route list (if any) comprise the extended route set and are usually the more expensive routes to the destination. Only routes in the initial route set should be defined to allow OHQ. OHQ will not be offered by routes in the extended route set even if they are defined to allow OHQ.

Calls which do not meet the preceding requirements for OHQ eligibility may be offered CBQ at this point.

OHQ availability

The OHQ software performs an "availability" test to prevent calls from entering the OHQ when there is a high probability that the call will timeout before a trunk becomes available. The following procedure is used to make this test.

- For each trunk route, a counter is maintained which reflects the current number of calls with the maximum queue priority of 3 queued against it. (This includes all calls in OHQ and those CBQ and Ring Again calls that are currently at priority 3.)
- Each trunk route has a threshold value which indicates the maximum number of priority 3 calls that can be queued against it before OHQ timeout becomes a high probability. Before a call is placed in the OHQ the current queue count is compared with the threshold value for each eligible trunk route in the initial set of routes. If at least one of the trunk routes has a count which is less than or equal to the threshold value, the call is allowed to OHQ against all OHQ eligible routes.

OHQ offer

If all eligibility and availability checks are successful, the call originator is given a distinctive OHQ offer tone (a 1s burst of 440 Hz tone) when the call enters the OHQ. The call originator either accepts the OHQ offer by remaining off-hook, or declines the offer by going on-hook.

OHQ calls are placed in a priority-ordered queue with all other calls waiting for trunks. OHQ calls are assigned maximum priority (3) because other network facilities may be held while the call is queued. A timer with an initial value equal to the software-defined OHQ time limit is started to limit the duration of the OHQ. (The OHQ time limit is service changeable within the range of 2 to 60 s.)

The queue is examined whenever a trunk becomes idle in a trunk route on which one or more calls are queued. If a call is found which can be terminated on an idle trunk, the available trunk is seized and the call terminated.

If the OHQ timer expires before the call can be terminated, the call is removed from the OHQ. At this time, the remaining eligible routes in the route list (the extended set) are examined and the call is either terminated or given standard blocking treatment if no facilities are available. (CBQ will not be offered at this point as OHQ was already offered.) Expensive Route Warning (ERWT) is not given to calls which have been queued even if terminated on expensive facilities.

OHQ can be offered to call originators located at a Node, Main, Conventional Main or ETN switch. Also, as a call progresses through the network, OHQ can be offered to the call originator from any of the Nodes or Mains that are used to process the call (i.e., OHQ can be offered more than once for a given call).

OHQ traffic measurements

Traffic measurement data related to usage of the OHQ feature is accumulated at a Node or Main that is equipped with Network Traffic (NTRF) feature. Data for OHQ usage is available in the publication *Traffic measurement* (553-2001-450).

Routing measurements (TFN001)

Available at a Node or Main. Reflects on a route list basis the number of calls placed in the OHQ, the average time a call waited in the OHQ and the number of calls abandoned while in the OHQ.

NCOS measurements (TFN002)

Available at a Node or Main. Reflects on a NCOS basis the number of call placed in the OHQ and the average time a call waited in the OHQ.

Incoming trunk group measurements (TFN003)

Available at a Node. (The measurements are also available at a Main if the OHQ feature is equipped and some incoming trunk groups are defined as OHQ eligible.) Reflects on an incoming trunk (or two-way) trunk group basis the number of calls placed in the OHQ and the average time a call waited in the OHQ.

Feature interactions

The OHQ feature has the following interactions with existing Meridian 1 features.

Call modification

Station users are not allowed to activate call modification features (hold, call transfer, conference) while waiting in the OHQ. Switchhook flash used to activate features from 500/2500 sets is ignored. Similarly, operation of SL-1 or digital set feature keys is ignored.

Camp-on, call waiting

If the attendant extends a call to a station that is in the OHQ, the call-waiting tone is not offered to the station. If the attendant releases, the call is camped-on the OHQ station but no warning tone is given. When the camp-on call is recalled to the attendant console, the attendant can repeat the camp-on procedure. Once the OHQ call is in an established state, the camp-on tone is provided.

Attendant functions

- Attendant cannot barge-in during trunk seizure for OHQ calls.
- If the attendant extends a network call for a station user and the call is offered OHQ, the attendant must inform the caller of the OHQ offer before releasing from the call.
- The attendant is not allowed to operate the release key or another loop key if the source call is in conference and the destination call is in the OHQ. Operation of the release destination key is permitted, however, and causes the OHQ call to be abandoned.

Call-Back Queuing

Call-Back Queuing (CBQ) is a Node feature which provides queuing for network calls which encounter busy or blocked facilities at the Node. CBQ enables the calling party to go on-hook after activation of the Ring Again feature [see *Features and services* (553-3001-305)] and receive a callback from a Node when a network facility becomes available.

The CBQ feature is available only to stations located at a Node. Unlike OHQ, CBQ is offered only at the originating Node. Access to CBQ is accomplished through the existing Ring Again feature.

CBQ options

Two options for CBQ eligibility are defined by the call originator's NCOS. The first option, CBQ(i), means the CBQ may be offered after only the initial route set of a route list has been examined for an available route. The second option, CBQ (a), means that CBQ may be offered after both the initial and extended (i.e. all) route sets of a route list have been examined. In either case, a call offered CBQ is queued initially against only the initial route set.

CBQ eligibility

Before offering CBQ to a call originator, the following eligibility tests are performed.

- At least one of the routes in the initial route set is defined as CBQ eligible.
- The user's NCOS is defined as permitting CBQ, either CBQ(i) or CBQ(a).

- The call is not eligible for OHQ. Calls that are eligible for both OHQ and CBQ will be offered OHQ.
- The user's telephone is allowed access to the Ring Again feature and does not have another CBQ or Ring Again call already in the queue.
- The CBQ feature is enabled for this customer.

CBQ(i) eligible

For call originations by a caller defined as CBQ(i) eligible, the system searches the initial route set for an available route. If no available route is found, CBQ is offered to the caller subject to the CBQA eligibility tests.

CBQ(a) eligible

For call originations by a caller defined as CBQ(a) eligible, the system examines the initial route set for an available route. If no available route is found, the extended route set is then searched for an available route. If an available route is not found in the extended set, then CBQ is offered, subject to the CBQ eligibility tests. However, if an available route in the extended route set is found that is designated as expensive, and the user's NCOS allows ERWT, the tone is given and the system delays terminating the call. During this delay the user has the following options.

- Refuse the expensive route by abandoning the call.
- Wait, and allow the call to complete over the expensive route.
- Activate the Ring Again feature (feature key or access code) to place the call in the CBQ. The user must be CBQ(a) eligible; otherwise, operation of the Ring Again feature is ignored.

CBQ offer

The CBQ offer consists of an optional recorded announcement followed by overflow tone. If the station user wishes to accept the CBQ offer, Ring Again must be activated within 30 s. Ring again activation follows present feature operation for SL-1 or digital and 500/2500 sets [see *Features and services* (553-3001-305)]. The CBQ offer can be refused by going on-hook. If the station user neither accepts nor refuses the CBQ offer within 30 s, the call is force disconnected.

CBQ calls are placed in a priority-ordered trunk queue (together with OHQ calls, if any) with a starting priority and maximum priority defined by the call originator's NCOS. At the same time, two timers are started – a queue promotion timer and a route advance timer – each with values defined through the originator's NCOS. At intervals defined by the queue promotion timer, the priority of the call is incremented until it reaches its maximum priority. Each time the call priority is incremented, its position in the CBQ is advanced. If the route advance timer reaches its maximum value before the call can be terminated on a route in the initial set, the extended set or routes is added to the routes that the call is currently queued against.

Expensive route warning tone is not given to calls which have been queued, even if terminated on expensive facilities. Unless cancelled by the call originator, CBQ calls remain in the queue until they have been offered a trunk; there is no time limit on CBQ calls.

CBQ callback

When a trunk becomes available for a CBQ call, it is seized to prevent incoming originations during the CBQ callback period. Outpulsing of digits (either those originally dialed by the user or those required as a result of digit manipulation) is started at a slow, fixed rate. The number of digits to be outpulsed determines how long the trunk can be held while CBQ callback is being offered to the originating station. The system computes this time by allowing 10 s before the first digit is outpulsed at 2.56 s between subsequent digits.

The originator of the call is alerted to the CBQ callback by either tone buzzing and winking ring again feature lamp (SL-1 or digital set) or short bursts of ringing (500/2500 set). The set user must accept the callback within the computed value of outpulse time or the service-changeable CBQ time limit of 10 to 30 s, whichever is less. A user with a 500/25000 set must accept the callback within 6 s. Acceptance of the CBQ callback is performed with present Ring Again operating procedures. (When a CBQ callback is answered at a digit display set, the original dialed digits are displayed.) If the user does not answer the callback within the time limit, the call is removed from the queue and discarded. If the user accepts the callback within the time limit, the call is terminated. A CBQ call can be cancelled by the originating station via the existing procedures for Ring Again cancellation.

CBQ traffic measurements

Traffic measurement data related to the usage of the CBQ feature is accumulated at a Node or Main equipped with the Network Traffic (NTRF) feature. Data for CBQ usage is available in the publication *Traffic measurement* (553-2001-450).

Routing measurements (TFN001)

Available at a Node or Main. Reflects on a route list basis the number of CBQ offers, the number of CBQ callbacks, the average time in the CBQ and the number of CBQ cancellations.

NCOS measurements (TFN002)

Available at a Node or Main. Reflects on an NCOS basis the number of CBQ calls and the average time in the CBQ.

Feature interactions

The CBQ feature interacts with the following features.

Barge-In, force disconnect

Between the time a trunk is seized for a CBQ call and the user accepts the CBQ callback, the trunk can be stolen by the attendant or force disconnected through service change. If this occurs, there is no guarantee that the call can be terminated when the user accepts the CBQ callback. Under these circumstances, the call is treated like a new origination and NARS/BARS is used to reattempt termination. This can result in the call being blocked and being offered CBQ a second time.

Hunting, call forward, multiple appearance DN

CBQ callbacks to stations at a Node are offered only to the originating station, regardless of the hunting or call forwarding which may be in effect. Other appearances of a station's Directory Number (DN) are not offered the callback.

Attendant functions

Because the Ring Again feature is not supported at attendant consoles, CBQ is not offered to an attendant regardless of the CBQ eligibility of the NCOS assigned to the attendant.

Coordinated Call-Back Queuing

The Coordinated Call-Back Queuing (CCBQ) feature enables stations at a Meridian 1 Main to be offered CBQ when network calls are blocked at the serving Meridian 1 Node. When facilities become available at the Node, the call originator at the Main is alerted by a callback from the Node. (This feature requires that Main and associated Node be equipped with the network signalling feature).

CCBQ eligibility

When a station at a Main originates a network call through a Node, the NCOS of the call originator, call type, and whether the station is allowed access to the ring again feature is transmitted to the Node. (If an authcode is entered at the Main prior to dialing a network call, the NCOS associated with the authcode is transmitted to the Node.) When received by the Node, this NCOS is used to determine CCBQ eligibility and is used for the duration of the call, unless further modified by the Authcode Conditionally Last feature [see *Basic and network authorization code description* (553-2751-103)].

The CBQ eligibility tests are performed. In addition, a check is made that the incoming trunk group from the Main is defined (at the Node) to permit CBQ and that the type allows CBQ. CCBQ is offered to the user at the Main if the eligibility tests are successful. If the tests are unsuccessful, standard call blocking is applied to the call.

As for stations at a Node, the call originator at a Main can invoke Ring Again upon receipt of ERWT if the originator's NCOS is defined at the Main as CBQ (a) eligible.

CCBQ offer

The CCBQ offer and acceptance sequences are identical to those for stations at the Node (SBQ offer). The optional recorded announcement and overflow tone are provided by the Node. The offer cannot be accepted until the recorded announcement is completed and the offer tone is started. In addition, after the recorded announcement is provided, the Node transmits a signal to the Main. This signal indicates that the call is in a state which allows Ring Again.

When the call originator at the Main activates ring again the Main assigns a unique "queue identification" number to the call. This number is transmitted to the Node to indicate CCBQ acceptance. At the Main, the call is placed in a holding queue. At the Node, the call (together with the queue identification number) is placed into the trunk queue. The Main to Node tie trunk is released.

CCBQ callback

When an outgoing trunk is seized by the Node for a CCBQ call, slow outpulsing is started to hold the trunk while a callback is made to the call originator at the Main. The Node seizes an available (note) tie trunk to the Main and transmits the "queue identification" number of the call to the Main. The Main the initiates a callback to the call originator. Callback presentation to the call originator is as for standard ring again (CBQ callback).

> *Note:* If no tie trunks to the Main are available, the outgoing trunk is released and can be offered another call. The CCBQ call retains its position in the queue but is not offered another trunk until a tie trunk to the Main becomes available.

When the call originator at the Main accepts the CCBQ callback, answer supervision is sent from the Main to the Node. The Node then completes the call.

If the call originator is equipped with a 500/2500 set and is engaged in a call when the Node initiates a CCBQ callback, a signal is transmitted from the Main to the Node. The Node releases the outgoing trunk and places the CCBQ call into a holding queue for 5 min. No attempt is made to seize another outgoing trunk for the call until the holding time expires. This process occurs only once.

If the originating station is still busy after the holding time has expired, the CCBQ is cancelled automatically at the Node. No indication is given to the call originator of the CCBQ cancellation.

To prevent the CCBQ call from remaining indefinitely in the holding queue at the Main, the Main sets a time limit of one hour for CCBQ calls. When this time limit expires, the CCBQ call is cancelled automatically. (CCBQ callback to a busy station is as for normal Ring Again).

The call originator at the Main can cancel the CCBQ call at any time. The Node is not aware of the cancellation until the CCBQ callback is attempted.

CCBQ traffic measurements

Traffic measurement data related to usage of the CCBQ feature are accumulated at the Node. The data is reflected in the Incoming Trunk Group Measurements (TFN003). The measurements reflect, on an incoming trunk group basis, the number of CCBQ offers, the number of CCBQ offers that are accepted and the number of CCBQ callbacks not answered or CCBQ cancellations. [Refer to *Traffic measurement* (553-2001-450).]

Feature interactions

The CCBQ feature interacts with the following Meridian 1 features.

Initialization

If the Main initializes while calls are queued at the Node, CCBQ callbacks from the Node are not answered because the initialization has cleared the holding queue at the Main. The Node treats these calls as callback-no-answer calls and cancels the CCBQ automatically. If the Node initializes, CCBQ calls in the trunk queue are lost. The Main cannot detect this situation. To prevent calls from remaining indefinitely in the holding queue at the Main, Main sets a time limit of one hour for CCBQ calls. If a call back from the Node is not received within one hour, the Main cancels the CCBQ calls automatically.

Attendant functions

Attendants at a Main are not offered CCBQ. Attendant barge-in on trunks involved in CCBQ operations results in cancellation of the CCBQ call.

AIOD and ANI

Automatic Identification of Outward Dial (AIOD) and Automatic Number Identification (ANI) facilities can be used to complete CCBQ calls from a Node. The outgoing toll call is billed to the access tie trunk rather than the station at the Main.

Coordinated Dialing Plan

If CCBQ is offered to a CDP call as a result of trunk blocking at the Node, the optional recorded announcement is not given before the CCBQ offer tone.

Call transfer, conference

Stations at a Main can be offered CCBQ when initiating transfer or conference calls. CCBQ is not offered to 500/2500 stations performing these operations.

Feature limitations

A station at a Main is only allowed 10–12 s to activate ring again after the CCBQ offer (or ERWT) rather than the 30 s allowed for stations at the Node. Accordingly, the time interval of the recorded announcement (if provided) must be properly engineered to ensure that the station at the Main has time to activate CCBQ when it is offered.

The time available to answer a CCBQ callback at stations at a Main is 2–3 s less than the time available to stations at the Node, due to the time required for access trunk seizure and signaling.

Tie trunk groups used to provide CCBQ to a Main cannot be arranged for loop start dial repeating signaling and cannot have joint far-end disconnect control as these arrangements do not allow the call to be disconnected after a CCBQ callback.

Tones and recorded announcements used for CCBQ to users at a Main are provided by the Node. Traffic loads imposed on these facilities at the Node must be considered when engineering Nodes.

Call-Back Queuing to Conventional Mains

The Call-Back Queuing to Conventional Mains (CBQCM) feature enables call originators at a Conventional Main to access the CBQ feature at a Node.

CBQCM eligibility

When a station at a Conventional Main originates a network call through a Node, the NCOS assigned to the incoming trunk group is used to determine the CBQCM eligibility. This NCOS, as well as the incoming trunk group, must be defined as CBQ eligible.

> *Note:* If the call originates from an ETN switch, the Traveling Class Mark (TCM) transmitted to the Node, by the ETN switch, must be compatible with a CBQ eligible Facility Restriction Level (FRL) at the serving Node.

CBQCM offer

The CBQCM offer to call originator at a Conventional Main consists of an optional recorded announcement, followed by special (interrupted) dial tone. (The announcement and tones are provided from the Node.) To accept the CBQCM offer, the call originator dials the extension number associated with the telephone being used for the call. When the last digit of the extension number is dialed, a confirmation tone (three 256 ms bursts of dial tone) is sent from the Node to the call originator. The call is placed in the CBQ at the node when the call originator goes on hook.

The CBQCM offer can be refused by going on-hook any time before the last digit of the extension number is dialed or by remaining off-hook for longer than 30 s after receipt of the confirmation tone. If the CBQCM is neither accepted nor rejected within 30 s, the caller is given overflow tone (from the Node) and the call is disconnected.

CBQCM callback

When an outgoing trunk becomes available at the Node, it is seized and slow outpulsing is started. The Node then seizes a tie trunk (See Note) to the conventional main and outpulses the extension number of the call originator. The call originator must answer the callback before slow outpulsing is completed; otherwise, the callback is cancelled and the outgoing trunk is released.

> *Note:* If no tie trunks are currently available to the conventional main, the Node releases the outgoing trunk. The CBQCM call retains its position in the queue but is not offered another outgoing trunk until a tie trunk to the Conventional Main becomes available.

When the call originator answers the CBQCM callback, answer supervision must be transmitted from the Conventional Main to the Node. Upon receipt of answer supervision from the Conventional Main, the Node transmits a tone (three 256 ms bursts of dial tone) to notify the call originator that the call is a CBQCM callback, and completes the call.

If the call originator's station is busy, or the originator does not answer when the callback is placed, the Node places the call in a suspended state for 5 min. After 5 min, another callback is attempted if the outgoing trunk is free. If the station which originated the call is still busy or does not answer, the Node cancels the call.

No provision is made for CBQCM cancellation by a call originator at a Conventional Main. Once the CBQCM offer is accepted, the call remains in the queue until the Node initiates a callback.

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SL-1

Network Queuing

Description

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SL-1

Coordinated Dialing Plan

Description

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Standard, release 1.0. Reissued for compliance with Northern Telecom standard 164.0 and to include updates for X11 release 15. Updates are indicated by revision marks in the margins.

December 1, 1991

This document is reissued to include technical content updates. Due to the extent of changes revision bars are omitted.

Contents

Feature description

The Coordinated Dialing Plan (CDP) feature enables a customer with a number of local Meridian 1s to coordinate the dialing plan for stations at these switches. When implemented, the Coordinated Dialing Plan (CDP) feature enables a station at one switch to call a station at another switch within the CDP group by dialing a unique 3 to 7 digit number, without access codes and associated pauses for dial tone. When equipped with the Directory Expansion (DNXP) package, this number can have up to 10 digits.

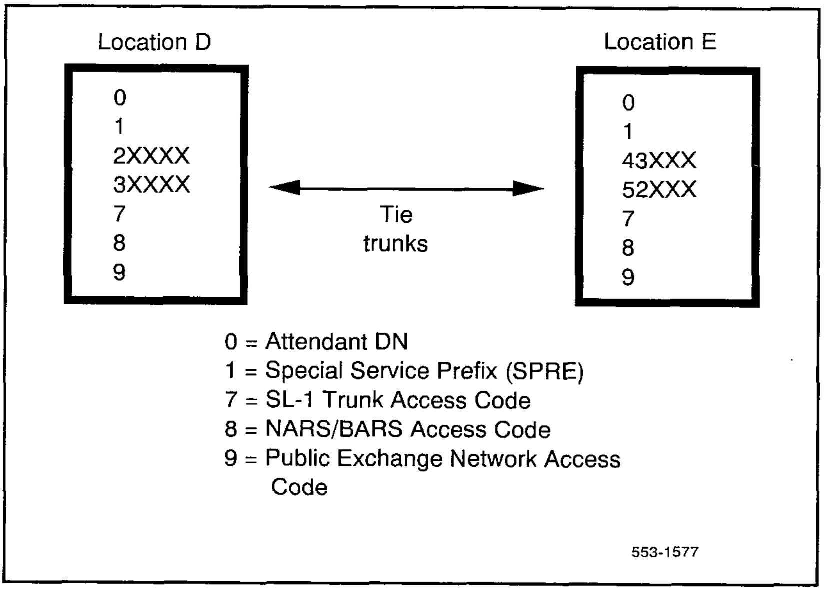
The CDP software provides the translation and digit manipulation capability required to implement the Coordinated Dialing Plan (CDP). Calls dialed within the CDP format can be terminated locally after digit translation or digit deletion. Alternatively, calls can be routed to a remote switch in the CDP group following digit translation, route selection, and digit deletion or insertion. Figure 1 illustrates how a coordinated dialing plan would be implemented at two customer locations.

Steering codes

In Figure 1, users at Location D can call stations at Location E by dialing 43XXX or 52XXX. Similarly, users at Location E can call stations at Location D by dialing 2XXXX or 3XXXX. If a user at Location D dials 43XXX or 52XXX to reach a station at Location E, Location D uses the digits "43" or "52" as a Distant Steering Code (DSC) to select the trunk group to Location E. Similarly, if a user at Location E dials 2XXXX or 3XXXX to reach a station at Location D, Location E uses the digit 2 or 3 as a Distant Steering Code (DSC).

The same format is used for calling local stations. For example, users at Location E dial 43XXX or 52XXX to reach local stations at Location E. In this case, the Meridian 1 interprets the digits 43 or 52 as a Local Steering Code (LSC) and deletes them from the dialed number in order to terminate the call locally.

Figure 1
Example of a Coordinated Dialing Plan



The maximum number of leading digits that can be deleted from a local steering code is 4. However, if the DNXP package (150) is equipped, up to 7 digits can be deleted from the Local Steering Code SPRE (LSC).

If the Meridian 1 at Location E is arranged to provide centralized access to the public exchange network, the digit 9 at Location E is a Trunk access code for public exchange access. At Location D, the digit 9 is a Trunk Steering Code (TSC) which uses digit manipulation to insert the required digits to route the call through Location E to the public exchange network.

The CDP feature supports up to 5000 steering codes. Steering codes can be composed of one, two, three or four digits. At each switch in the CDP group, the steering codes must be distinct from any other assigned DNs codes. As Figure 1 shows, 0 is reserved as the attendant access code; 1 is reserved as the Special Service Prefix (SPRE); 7 is reserved as a Meridian 1 trunk access code; 8 is reserved as a Basic Alternate Route Selection / Network Alternate Route Selection (BARS/NARS) access code; and 9 is reserved as the public exchange network access code. This means there are five digits remaining that can be used as the leading digits of steering codes – 2, 3, 4, 5, and 6. Switch D chooses 2 and 3; switch E uses 4 and 5.

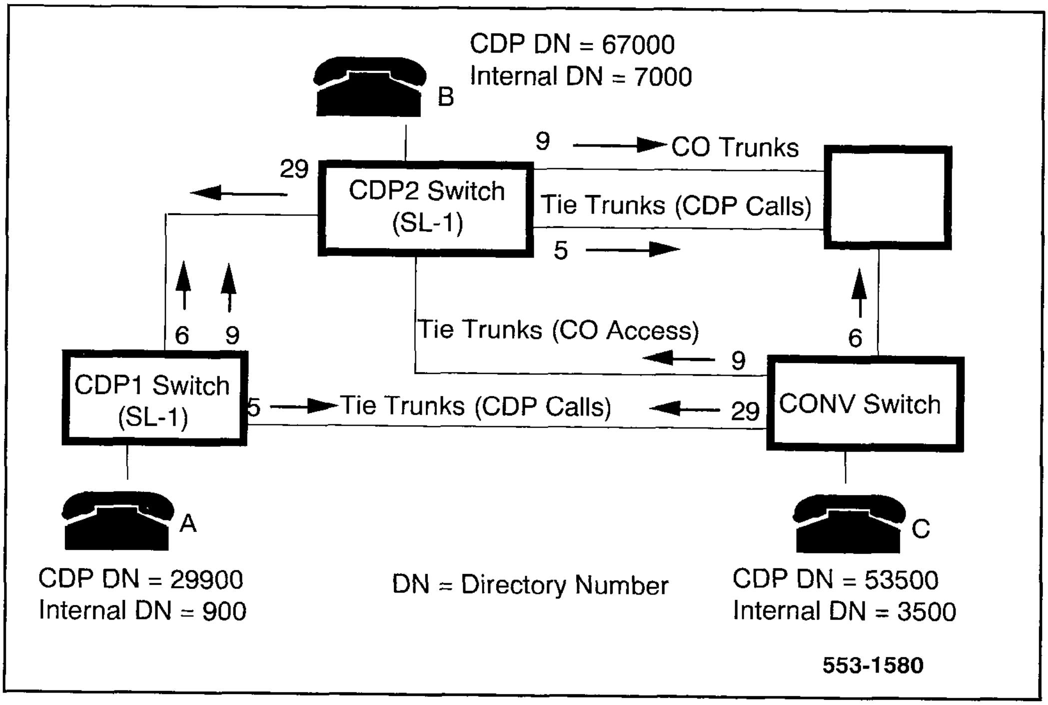
With Release 13 and later, the CDP feature can support up to 10,000 steering codes. If the DNXP package is equipped, these steering codes can be up to 7 digits.

A CDP Directory Number (DN) consists of an internal DN prefixed with the appropriate steering code. The CDP DN is allowed up to 7 digits maximum; but, if the DNXP package is equipped, this number can be increased to 10 digits maximum. A typical CDP configuration is shown in Figure 2.

Conventional Switch Access

If a conventional (CONV) switch without the CDP software is integrated as part of a CDP group (Figure 2), the steering codes defined at a CDP switch to access the conventional switch can be inserted or deleted by the CDP switch. The steering codes are inserted if the conventional switch is identified by more than one steering code; they are deleted if all the station numbers at the conventional switch begin with the same steering code.

Figure 2
A typical CDP configuration



Calls to a CDP switch from the conventional switch are made by dialing the desired CDP DN. The CONV switch uses the digit 6 as a trunk access code for the Tie trunk route to switch CDP2. After Tie trunk seizure, the CONV switch outpulses the remaining digits (7000) to CDP2. At CDP2, the digit 6 is inserted on the incoming Tie trunk from the CONV switch, prior to receipt of any digits from the CONV switch, and the call is completed to station E.

Local calls at the CONV switch are made by dialing only the internal DN (3500), rather than the CDP DN (53500), unless the CONV switch can be arranged to absorb the digit 5 or is based on a 5-digit numbering plan.

As shown in Figure 2, switch CDP2 is arranged to provide centralized access to the public exchange network. For users at the CONV switch to access this capability, a separate Tie trunk route must be provided to switch CDP2. This is because switch CDP2 is arranged to insert the digit "6" on the incoming Tie trunk route from the CONV switch used for CDP calls. For public exchange network calls, the digit 9 must be inserted on the incoming Tie trunk route from the CONV switch. Similarly, if users at the CONV switch are to be allowed access to the ESN capabilities (BARS/NARS) at switch CDP2, another Tie trunk route must be provided for this purpose.

Network Class of Service

Network Class of Service (NCOS) is an integral part of the CDP feature. NCOS provides the means to control the following.

- which trunk routes are eligible to be accessed for completion of a CDP call
- whether or not queuing is offered to the call originator
- whether or not the originator of a CDP call receives an Expensive Route Warning Tone (ERWT) when an expensive trunk route is selected to complete the call.

A CDP equipped switch can accommodate four NCOS groups (0-3), each group with different route-access characteristics. See Figure 3 if the BARS/NARS feature is also equipped. Once each NCOS group is defined through service change; then line, trunk, and attendant groups are assigned to the NCOS group which best serves their requirements. The NCOS group to which each line, trunk or attendant group is assigned is independent of the regular Class Of Service assigned to them.

With Release 13 and later, a CDP equipped switch can accommodate 100 NCOS groups (0-99) whether it is equipped with BARS/NARS, or the New Flexible Code Restriction (NFCR).

Compatibility with ETN switches

The Traveling Class Of Service (TCOS) is equivalent to the Traveling Class Mark (TCM) used at Electronic Tandem Network (ETN) switches. It provides a mechanism for the system to control route access (FRL) and off-hook queuing (OHQ) eligibility for calls placed to or through another Node, or ESN Main, and enables the switch to interface with ETN switches.

When a Distant Steering Code (DSC) call is made from an Electronically Switched Network (ESN) Node to an ETN switch, the dialed digits, together with the TCOS number (0-7), are sent to the connected ETN switch. Similarly, when a DSC call is made from an ETN switch to an ESN Node, the dialed digits, together with the TCM number (0-7), are sent to the connected ESN Node. On a tandem connection to the ESN Node interprets the received TCM as a TCOS number. The received TCM replaces the FRL of the NCOS assigned to the incoming trunk group from the ETN switch.

Assumptions

The assumptions are as follows:

- Only DSC, not Trunk Steering Code (TSC), calls are supported with this capability.
- When a DSC call is terminated on a switch as a Local Steering Code (LSC) call, the transmitted TCOS/TCM number from the connected ETN switch is not collected and saved by the terminating switch.

Facility Restriction Level

Included as part of each NCOS group is a Facility Restriction Level (FRL) number which ranges from 0 (low-privilege) to 7 (high-privilege). The FRL is used by the CDP software to determine the alternate route selection choices available for CDP call attempts by users within an NCOS group.

Example

A station user assigned in an NCOS group having an FRL of 3 would be allowed access only to alternate route selection choices assigned an FRL of 3 or less; access to trunks with an FRL greater than 3 would be denied.

Routing

Thirty-two route lists (0-31) can be defined at a switch equipped with CDP. See Figure 3 for other parameters if CDP is equipped with Basic Alternate Route Selection or Network Alternate Route Selection. A route list is used to define the alternate route choices for CDP calls to a particular destination. Route choices in a route list are called route list entries. There can be up to three (0-2) route list entries associated with each route list, or seven (0-6) in Release 13 and later.

Route lists are associated with each Distant Steering Code (DSC) and Trunk Steering Code (TSC) that can be dialed at a CDP switch. Local Steering Codes (LSC) are not associated with route lists. Each code is defined to the CDP software, together with the number of the route list that must be accessed for call completion to the destination indicated by the steering code. The entries in the specified route list are than searched sequentially for an available and eligible trunk route.

Release 15 and later software allows CDP to route Direct Inward Dialed (DID) calls over CO and WATS trunks using a Distant Steering Code (DSC). The feature is controlled by an option defined in the Customer Data Block (LD 15) found in the *X11 Input/output guide* (553-3001-400). This enhancement applies to CO, WATS, DTI and ISDN type trunks.

Digit manipulation

Route list entries can be associated with digit manipulation tables. There can be 32 (0-31) digit manipulation tables defined at a CDP switch. See Figure 3 if BARS/NARS is also equipped. Every digit manipulation table except 0 can be defined to delete up to 15 digits from a dialed CDP number, and to insert up to 24 leading digits, including the asterisk. Digit manipulation table 0 is used as an indication to the CDP software that no digit manipulation is required.

Time of day schedules

Two (0-1) time of day (TOD) schedules can be defined at a CDP switch. See Figure 3 if BARS/NARS is also equipped. Each route list entry is associated with a TOD schedule. When a route list entry is selected for a CDP call, the CDP software compares the current time with the TOD schedule assigned to the route list entry. If the current time is within the schedule, the route list entry is used for the call. If the current time is not in the schedule or, if the TOD schedule is turned OFF, the route list entry is not used for the call. Each TOD schedule can be turned ON or OFF by the customer through service change.

Queuing

Queuing against local stations is provided by the standard Ring Again (RGA) feature. Please refer to the *X11 Features and services description* (553-3001-305). For calls directed to a remote CDP switch, Ring Again can be applied if all local outgoing trunk routes to the remote CDP switch are busy or blocked. Ring Again cannot be applied against busy or blocked telephones, or consoles at the remote CDP switch. Trunks can only be rung again if CCBQ or CBQM are equipped. Intercept treatment is not provided until the full CDP number (or trunk steering code) is dialed.

For local and network queuing descriptions, refer to *Basic and Network Alternate Route Selection* (553-2751-100). For ESN operations in an ISDN environment, consult the *ISDN Primary Rate Access description and administration* (553-2901-100).

CDP traffic measurements

Traffic measurement data related to CDP feature usage is available at an Meridian 1 equipped with the Network Traffic (NTRF) feature. The user should refer to *Traffic Measurement* (553-2001-450).

Figure 3
Summary of CDP parameters

Parameter	CDP stand-alone	CDP with BARS	CDP with NARS
Network Class of Service Groups (Note 2)	0-3 0-(99)	0-7 0-(99)	0-15 0-(99)
Facility Restriction Levels	0-7	0-7	0-7
Time-of-Day schedules	0-1	0-7	0-7
Digit Manipulation tables	1-31	1-255	1-255
Route lists	0-31 0-127	0-127	0-255
Route list entries	0-2 0-(6)	0-7 0-31* 0-63**	0-7 0-31* 0-63**
Supplemental Digit Restriction tables	—	0-255	0-512
Steering codes	1-5000 1-(10000)	1-5000 1-(10000)	1-5000 1-(10000)

Note 1: Numbers preceding ** apply ti X11 release 15 and greater. Numbers with * apply to X11 releases 13 and 14.

Note 2: The BARS/NARS features are described in detail in the *Basic and Network Alternate Route Selection general description* (553-2751-100).

Note 3: If New Flexible Code Restriction (NFCR) is equipped in conjunction with CDP, the number of available NCOS groups is 8; with Release 13 and later, this number is increased to 100.

>P feature can be implemented at a switch equipped
S software features. If such is the case, the following

r CDP calls must be distinct from the assigned
cess codes.

n be integrated with the NARS Uniform Dialing Plan
ple, a five-digit CDP number can be the same as the
a seven-digit UDP number.

ute lists, digit manipulation tables and TOD schedules
CDP calls.

the Off-Hook Queuing (OHQ) and Call-Back
features can use them when placing CDP calls.

a Screening (FCAS) does not apply to CDP calls.

can be applied to CDP calls. Please refer to *Basic and*
te Route Selection general description (553-2751-100)

made to the public exchange network when the
ion of Outward Dialing (AIOD) or Automatic Number
eature is equipped will have either the internal DN
iginates at the CDP switch interfacing to the public
access code if the call originates at another CDP

If a user at a local CDP switch calls the local
ser's internal DN (not the full CDP DN) is displayed. If
h calls an attendant at another CDP switch, the trunk
ber number of the incoming trunk are displayed.

ant features are supported at a local CDP switch but are
n CDP switches:

recall

rify

ing

git display set handles CDP calls.

lete dialed CDP DN is displayed at

access code and member number of
yed.

nating set, the complete dialed CDP
or is picked up by another station, the
ion is displayed. At the terminating
ating set is displayed.

R interacts with CDP calls in the same
e *Basic and Network Alternate Route*
ll description of NXFER.

SL-1

Coordinated Dialing Plan

Description

Release 2.0
Standard
December 1, 1991
Printed in U.S.A.

northern
telecom

SL-1

Basic and Network Authorization Code

Description

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Revision history

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December 1, 1991

This document is reissued to include technical content updates. Due to the extent of changes revision bars are omitted.

August 1, 1993

This document is reissued for updates and changes resulting from X11 release 19. All updates are noted with revision bars in the margins.

Contents

List of Tables

Introduction

The Basic and Network Authorization Code features enable selected users to temporarily override the access restrictions assigned to a station or trunk. A user can enter an authorization code (authcode) to access more of the system facilities than would normally be allowed to the particular station or trunk because of the assigned Network Class of Service (NCOS), Class of Service (COS), and Trunk Group Access Restriction (TGAR) codes.

These features are useful when a user initiates a call from someone else's telephone and requires access to more system facilities (such as access to long distance calling) than are allowed to that telephone. Entering a valid authorization code enables the user to access these additional features. After a valid authorization code is entered, the NCOS, COS, and TGAR associated with the authorization code replace the NCOS, COS, and TGAR associated with the telephone for the duration of the call.

Station Specific Authcode is a special feature available with X11 release 19 that enables the system administrator to control the level of authorization code access on a per telephone basis.

Document overview

This publication describes the authorization code features as offered on X11 software:

— Basic Authorization Code (BAUT) for general applications

— Network Authorization Code (NAUT) for network applications

Other documentation

Other Northern Telecom Publications (NTPs) related to BAUT and NAUT are:

Basic and Network Alternate Route Selection description (553-2751-100)

Coordinated Dialing Plan description (553-2751-102)

Feature description

Basic Authorization Code

Basic Authorization Code (BAUT) (package 25) provides for up to 4096 authorization codes of 1 to 14 digits. Users can enter an authorization code after dialing the Special Prefix (SPRE) and the digit "6" before dialing any call, including a Network Alternate Route Selection (NARS), Basic Alternate Route Selection (BARS), or Coordinated Dialing Plan (CDP) call. With the BAUT feature, an authorization code can be entered when:

- originating a call from a local station or Tie trunk
- initiating a call transfer or conference from a local station
- originating a call via the Direct Inward System Access (DISA) feature

Note: Refer to *Basic and Network Alternate Route Selection description* (553-2751-100) for a description of the NARS and BARS features. Refer to *Coordinated Dialing Plan description* (553-2751-102) for a description of the CDP feature.

Authorization code validation

The software validates an entered authorization code on the basis of the number of digits dialed and the dialed digits themselves. If the number of digits in the entered authorization code does not match the defined authorization code length (authorization code Data Block, AUB, LD88), the authorization code is deemed invalid. Similarly, if the dialed authorization code digits are not defined in the authorization code table (AUT, LD88), the authorization code is deemed invalid.

When an invalid authorization code is encountered, no response is given to the user until the End-of-Dialing (EOD) timer expires. (This increases the security of authorization codes by making it difficult for an unauthorized user to determine the length of a valid authorization code.) When the EOD timer expires, overflow tone is given for 15 seconds and the call is force disconnected.

Station Specific Authcode

With X11 release 19 and later, Station Specific Authcode (SSAU), package 229, enables the system administrator to define the authorization code access level for each telephone. This feature applies to 500/2500 and digital telephones, and is implemented on a per telephone basis. It does not apply to BRI telephones.

Station Specific Authcode provides three levels of authorization code access:

1. AUThcode Unrestricted (AUTU)
   A telephone configured as AUTU has no authorization code access limitations.
2. AUThcode Restricted (AUTR)
   A telephone configured as AUTR can enter up to six assigned authorization codes. (The same authorization code may be assigned to more than one AUTR telephone.)
3. AUThcode Denied (AUTD)
   A telephone configured as AUTD has no access to authorization codes. Any authorization code entered will be rejected and the call will not be completed.

Operating parameters

The same authorization code may be assigned to more than one AUTR telephone.

There is cross-checking between LD10 and LD11, which define a station specific authorization code, and LD88, which ensures that the user has entered a valid authorization code.

LD88, which deletes an existing authorization code, does not check if the authorization code is assigned as a station specific authorization code before the deletion.

Attendant Administration does not support the assignment of Station Specific Authcode.

Feature operation

After an authorization code is entered, the Station Specific Authcode feature determines if the telephone is allowed to use the entered code. If the authorization code is not allowed on that telephone, the existing invalid authorization code treatment occurs. Otherwise, normal authorization code processing occurs.

Authorization code administration

Classcodes

With the NAUT and BAUT features, a “classcode” structure is part of authorization code administration. A classcode is a combination of COS, TGAR, and NCOS codes. There can be up to 116 (0-115) classcodes defined through the Authorization Code Data Block (AUB, LD88), each with a different combination of COS, TGAR, and NCOS codes. Authorization codes that have the same combination of COS, TGAR, and NCOS codes are assigned the same classcode.

Creating authorization codes

When creating new authorization codes, a classcode with which the new authorization codes are associated is specified. The new authorization codes will then be automatically assigned the COS, TGAR, and NCOS codes associated with the specified classcode.

> *Note:* The BAUT feature does not support automatic generation of authorization codes. With the NAUT feature, authorization codes can be defined individually by the customer or generated automatically by the Meridian 1.

Exemptcode

When an authorization code is to be removed from use, a facility exists to prevent that authorization code from being reused (i.e., the authorization code will not be accepted as valid input when individually defining authorization codes). This is accomplished through an "exemptcode." When an authorization code is removed from use, an "exemptcode" is assigned to the authorization code in place of the classcode. The "exemptcode" is the month (e.g., JAN, FEB, etc.) taken from the system clock. If an "exemptcode" is not requested, the removed authorization code is returned to the pool of unused authorization codes and can be reused at any time.

Default Facility Restriction Level

The Route List Block (RLB) program (LD86) is used to define a minimum Facility Restriction Level (FRL) for each route list. This minimum FRL (range 0-7) is used to determine whether or not to prompt for an authorization code entry after a call. If a minimum FRL is not specified, the actual minimum FRL in the initial route set is used as a default. Similarly, the Route Data block (RDB) program (LD16) is used to define whether to prompt for an authorization code entry on calls on incoming or two-way Tie trunk groups.

Network Authorization Code

The Network Authorization Code (NAUT) feature provides for up to 20,000 authorization codes of 1 to 7 digits.

> *Note:* With X11 release 13 and later, the authorization codes can be of 1 to 14 digits.

The NAUT feature incorporates all the features of the BAUT feature, adding two enhancements:

– a "conditionally last" option for entering an authorization code after dialing a NARS/BARS/CDP call

– allowing the attendant to enter an authorization code.

Authorization code conditionally last

With the NAUT feature, users can be prompted "conditionally" for an authorization code after dialing a NARS, BARS, or CDP call. The prompt is by an "authorization code request tone" which consists of 10 bursts of dial tone, followed by steady dial tone. (The authorization code request tone can, optionally, be preceded with an appropriate recorded announcement.) The user is prompted for an authorization code entry only if:

- an authorization code was not previously entered
- the Facility Restriction Level (FRL) associated with the user's Network Class of Service (NCOS) is less than the service change assigned minimum FRL of the route list that NARS/BARS/CDP would use for the call.

Users at a remote switch (Meridian 1 Main or Conventional Main) connected via Tie trunks to a Meridian 1 Node can (optionally) be prompted for an authorization code entry after dialing a NARS/BARS/CDP call. The user is prompted for an authorization code entry only if:

- an authorization code was not previously entered
- the FRL associated with the NCOS of the incoming (or two-way) Tie trunk is less than the minimum FRL of the route list that NARS/BARS/CDP would use for the call
- the route is defined in the Route Data Block (RDB), LD16, to prompt for an authorization code entry on incoming NARS/BARS/CDP calls

Users accessing a Meridian 1 Node via the Direct Inward System Access (DISA) feature to make a NARS/BARS/CDP call are prompted for an authorization code entry only if:

- an authorization code was not previously entered
- the FRL of the NCOS assigned to the DISA Directory Number (DN) is less than the minimum FRL of the route list that NARS/BARS/CDP would use for the call

Attendant input of authorization code

Normally, because an attendant is not restricted from accessing any system resource, there is no need for the attendant to have an authorization code. The Network Authorization Code feature enables the attendant to enter an authorization code for other callers. For example, the attendant can enter an authorization code (after dialing the SPRE and the digit "6") and complete a long distance call for a local station user whose COS is toll denied (TLD). If the Call Detail Recording (CDR) of authorization codes is defined for the customer, the local station user's authorization code digits appear in the CDR record for billing purposes.

Attendants are normally assigned an NCOS having a high FRL so that they can make any type of call, including NARS, BARS, or CDP calls. An attendant can, however, be prompted for an authorization code entry if the FRL required to access a route list for a NARS/BARS/CDP call is greater than the FRL of the attendant's NCOS.

Operating parameters

Users on PBX or Centrex systems connected via Tie trunks to a Meridian 1 Node can use the authcode conditionally last feature, provided that these systems transmit or repeat all digits dialed by the users in response to the authcode request. This feature cannot be used by certain systems that operate in senderized mode. Correct operation may require adjustment of EOD time-out on systems that employ simulated cut-through operation.

In a private network consisting of multiple switches equipped with the Authcode feature, authcodes should be requested only once on a given call. This requires careful engineering of:

— the Tie trunk group option for authcode prompting

— the minimum FRL values assigned to route lists

In a private network, users at a switch arranged for the Uniform Dialing Plan (UDP) via a dedicated trunk group to a Node can use the authcode conditionally last feature at the Node in the same manner as those stations located directly at the Node. However, these users cannot access the authcode after SPRE feature via the same trunk group.

Feature interactions

Feature key operations

While a user is entering an authcode, the following feature keys operate as intended and do not affect operation of the authorization code feature:

— Make Set Busy

— Buzz

— Volume Control

The operation of the following keys is ignored during authcode operation:

— Conference

— Override

— Call Forward and Call Transfer

— Call Pickup

— Charge Account

— Calling Party Number

— Privacy Release

— Ring Again

— Barge-In and Busy Verify

— Speed Call

— Recall

— Do Not Disturb

— Digit Display

The following key operations abort the authcode operation and any digits entered in authcode are ignored:

- Directory Number
- Paging
- Voice Call
- Not Ready
- In-Calls
- Call Waiting
- Hold
- Release

If the caller initiates a switchhook flash while entering an authcode, the results are unpredictable; the switchhook flash may be ignored or interpreted as the digit "1."

Authcodes after SPRE can be stored as speed call or autodial entries. When this is done, the stored number (entry) must contain only the access code and authcode digits. All digits in the entry after the access code are interpreted as authcode digits.

In the case of authcode conditionally last, authcodes can be stored as auto-dial entries but not speed call entries. If necessary, the caller can continue to enter more authcode digits after operation of the auto-dial or speed call key. However, for security reasons, authcodes should not be stored as auto-dial or speed call entries.

Call Detail Recording

If the CDR recording of authcodes is specified, a record is generated on the CDR device each time an authcode is entered. The record is passed to CDR only if one of the following occurs:

- The call becomes established (for example, a trunk is seized or local telephone answers).
- The call cannot be completed (for example, when no trunks are available).
- The Ring Again feature is applied to the call.

Authcode input via Tie trunks

Authcodes can be entered via access Tie trunks. Incoming or two-way Tie trunk groups at a switch equipped with the Network Authorization Code feature can be defined to prompt the user for an authcode entry.

Direct Inward System Access

If a caller makes a NARS/BARS/CDP call in association with a valid DISA call, the NCOS associated with the DISA DN is used for NARS/BARS/CDP route selection. If the FRL of this NCOS is too low to access the route list that NARS/BARS/CDP has selected for the call, the caller will be prompted for an authcode entry, unless an authcode (for example, Authcode after SPRE) was entered previously.

Barge-In/Busy Verify

If the attendant uses Barge-In or Busy Verify to break into a connection where an authcode is being entered, the authcode entry will be affected. If the code entered is invalid as a result, the user will be given overflow tone when the EOD timer expires.

Centralized Attendant Service

The Centralized Attendant Service (CAS) feature enables several remote switches to share the attendant services at one central location. See *Centralized Attendant Service description and engineering* (553-2681-100). A CAS attendant can enter an authcode via a Release Link Trunk (RLT), before connecting or transferring calls to the connecting remote PBX. If the CAS attendant enters a NARS/BARS/CDP number via an RLT, the NCOS associated with the attendant (at the remote PBX) is used in the NARS/BARS/CDP route selection process. If the FRL of this NCOS is inadequate, the CAS attendant may be prompted for an authcode entry.

Call Forwarding

The Call Forwarding feature provides two customer options: Call Forwarding-Originating Party's COS (CFO) or Call Forwarding-Forwarding Party's COS (CFF). With the NAUT feature and the CFO option, a caller may be prompted for an authcode entry after a call to a station which forwards the call to a NARS/BARS/CDP number. With the CFF option, the user will not be prompted by the local switch for an authcode entry after such a call.

Network Class of Service

An authcode entry modifies the user's NCOS for the duration of the call. The FRL associated with the user's assigned NCOS is used to determine if it is necessary to prompt for an authcode entry. After an authcode is collected and validated, the NCOS associated with the authcode is used for the duration of the call.

Network/Basic Alternate Route Selection

During NARS/BARS route selection, the FRL associated with the call originator's NCOS is compared with the FRL of the selected route list. If the originator's FRL is lower and no authcode was entered previously, the system may prompt for an authcode entry. A valid authcode modifies the originator's NCOS and, hence, FRL. This new FRL is then used for route selection.

Network Queuing

When an authcode is entered, the NCOS associated with the authcode is used to determine Network Queuing capabilities.

Coordinated Dialing Plan

Authcode after SPRE can be used before dialing a CDP call. If the NAUT feature is equipped, the “conditionally last” request for an authcode entry applies.

Implementation

This section describes the implementation steps for Basic and Network Authorization Codes. Refer to the *X11 input/output guide* (553-3001-400) for a complete description of these procedures.

The following responses to prompts in LD88 are required. Default responses appear in parentheses.

Table 1
Authcode data block (AUB) (Part 1 of 2)

Prompt	Response	Comments
REQ	NEW, CHG, PRT	Action request (create, modify, or print)
TYPE	AUB	Authcode data block
CUST	0-99	Customer number
SPWD	xxxx	Secure data password
ALEN	1-14	Number of digits in authcode
ACDR	Yes, No	Activate CDR for authcodes
RANR	0-511	RAN route number for 'Authcode Last' prompt (NAUT)
CLAS	(0)-115	Classcode value assigned to authcode (NAUT)
COS	aaa	Class of Service
TGAR	(0)-31	Trunk Group Access Restrictions

Table 1
Authcode data block (AUB) (Part 2 of 2)

Prompt	Response	Comments
NCOS	(0)-99	Network Class of Service
AUTO	Yes, No	Automatically generate authcodes
_SECR	0-9999	Security password (NAUT)
_NMBR	1-9999	Number of authcodes to be generated automatically (NAUT)
_CLAS	(0)-115	Classcode value assigned to authcode (NAUT)

Table 2
Delete Authcode data block

Prompt	Response	Comments
REQ	OUT	Action request (remove data)
TYPE	AUB	Authcode data block
CUST	0-99	Customer number
SPWD	xxxx	Secure data password
CODE	xxxx	Authcode (number of digits must equal ALEN)
CLAS	(0)-115	Classcode value assigned to authcode (NAUT)

Table 3
Authcode table entries

Prompt	Response	Comments
REQ	NEW, CHG, PRT	Action request (create, modify, or print)
TYPE	AUT	Authcode entries
CUST	0-99	Customer number
SPWD	xxxx	Secure data password
CODE	xxxx	Authcode (number of digits must equal ALEN)
CLAS	(0)-115	Classcode value assigned to authcode (NAUT)

Table 4
Delete Authcode table entries

Prompt	Response	Comments
REQ	OUT	Action request (remove data)
TYPE	AUT	Authcode entries
CUST	0-99	Customer number
SPWD	xxxx	Secure data password
CODE	xxxx	Authcode (number of digits must equal ALEN)
SECR	0-9999	Security password (NAUT)

To activate or deactivate Station Specific Authorization Codes at a particular telephone, you use LD 10/11:

Table 5
LD10/11 - Activate SSAU

Prompt	Response	Comment
REQ	NEW/CHG	Add or modify
TYPE	xxxx	Telephone type: 500 (500 or 2500) 2006, 2008, 2009, 2016, 2018, 2112, 2216, 2317, 2616, 3000, SL1
CLS	(AUTU), AUTR, AUTD	Authcode unrestricted Authcode restricted Authcode denied
MAUT	YES/(NO)	Modify assigned authcodes for this telephone
SPWD	xxxx	Correct security password (if one is defined)
AUTH	x nnnn	x is in the range of 1-6; nnnn is the assigned authcode (a valid authorization code defined in Overlay 88).
	X	Entering an uppercase X deletes an assigned authcode.
Note: Changing an AUTR telephone to AUTU or AUTD clears all assigned authcode information previously defined for that telephone.		

Table 6
LD20 - Station print

Prompt	Response	Comment
REQ	PRT	Print command
TYPE	xxxx	Type of Terminal Block
TN	l s c u	Terminal number
CDEN	xx	Card density
CUST	xx	Customer number
SPWD	xxxx	Security data password

Note: SPWD is not prompted if any of the following is true:

- — The Station Specific Authcode package (220) is not equipped
- — The response to the TYPE prompt is not TNB, SL1, 2000, 2003, 2009, 2018, 2112, 2317, 3000, ARIES, 2006, 2008, 2016, 2216, or 500
- — The response to the TN is more than one specific TN
- — The response to the TN prompt is a unique TN but the customer of this TN does not have a security data password defined
- — The response to the CUST prompt is not a specific customer
- — The response to the CUST prompt is a specific customer number, but the customer does not have a security data password defined

Table 7
LD81 - Feature print

Prompt	Response	Comment
REQ	1ST	List sets with the feature specified
	CNT	Count sets with the feature specified
CUST	xx	Customer number
FEAT	AUTU	Authcode unrestricted
	AUTR	Authcode restricted
	AUTD	Authcode denied

Feature operation

Authcode after SPRE

500/2500/SL-1 or digital telephones

To enter an authcode after the Special Prefix (SPRE), the caller proceeds as follows.

— If there is no call in progress, go off-hook or press a Directory Number (DN) key. If there is a call in progress, switchhook flash (500/2500 telephone) or press the call transfer or conference key (SL-1 or digital telephone) to obtain special (interrupted) dial tone.

— Dial the authcode access number (SPRE and the digit "6"). Dial tone is removed after the SPRE digit is dialed.

— Dial the authcode digits. A second dial tone is heard if the authcode is valid. If the authcode is invalid, no response is given for 30 seconds. Then the overflow tone is given for 15 seconds and the call is force disconnected. For more information about how invalid authcodes are handled, refer to "Invalid authcodes" on page 24.

— When the second dial tone is heard, dial the call in the normal manner. If call transfer/conference is in effect, complete the transfer/conference as normal.

Attendant

To enter an authcode after SPRE, the attendant proceeds as follows:

- If there is a call on the source loop, proceed to the next step. If there is no call on the source loop, press an idle loop (LPK) key.
- Dial the authcode access number (SPRE and the digit "6"), followed by the authcode.
- Dial as usual after receiving dial tone denoting a valid authorization code. (If the code is invalid, overflow tone is returned immediately.)

Authcode conditionally last

The following procedure is used to enter an authcode conditionally last from a 500/2500/SL-1 or digital telephone or attendant console (NAUT feature only).

- Dial a NARS/BARS/CDP call.
- Receive an "authcode request tone" (10 bursts of dial tone, followed by steady dial tone), optionally preceded with a recorded announcement, indicating that an authcode entry is required.
- Dial the authcode. The dial tone is removed after the first digit is dialed. If the authcode is valid, the call is processed as a normal call. If the authcode is invalid, no response is given for 30 seconds. Then the overflow tone is given for 15 seconds and the call is force disconnected. For more information about how invalid authcodes are handled, refer to "Invalid authcodes" below.

Invalid authcodes

When an invalid authcode is entered, the reorder tone occurs after the interdigit time out. (Interdigit timeout is set by prompts DIDT or DIND in LD15.) The authcode feature does not give the overflow tone immediately upon detecting an invalid authcode in order to prevent repetitive attempts. However, the network authcode feature (with the NAUT package) does return the overflow tone to the local originating attendant after the attendant enters an invalid authcode with the correct number of digits.

On a Tie trunk, entering an invalid authcode locks out the line.

Packaging

Basic Authorization Code is available as package 25.

Network Authorization Code is available as package 63.

Station Specific Authorization Codes is available as package 229. It requires package 25.

List of Terms

AUTD

Authcode denied class of service

AUTR

Authcode restricted class of service

AUTU

Authcode unrestricted class of service

BARS

Basic Alternate Route Selection

BAUT

Basic Authorization Code

CAS

Centralized Attendant Service

CDP

Coordinated Dialing Plan

CDR

Call Detail Recording

CFO

Call Forwarding-Originating

CFF

Call Forwarding-Forwarding

COS

Class of Service

DISA

Direct Inward System Access

DN

Directory Number

EOD

End-of-Dialing

FRL

Facility Restriction Level

NARS

Network Alternate Route Selection

NAUT

Network Authorization Code

NCOS

Network Class of Service

RDB

Route Data Block

RLT

Release Link Trunk

SSAU

Station Specific Authorization Code

SSP

Special Service Prefix

TGAR

Trunk Group Access Restriction

TLD

Toll Denied class of service

SL-1

Basic and Network Authorization Code

Description

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SL-1

Digital Trunk Interface/ Computer-to-PBX Interface

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Date: August 1, 1993

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August 1, 1993

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Contents

List of Figures

List of Tables

Introduction

Document overview

This document describes the Digital Trunk Interface (DTI) feature:

— functions

— operations

— associated hardware

DTI is an optional feature and is available on network enhanced systems equipped with X11 release 5 and later software.

Other documentation

Other Northern Telecom Publications (NTPs) related to DTI are:

- *Digital Trunk Interface/Computer-to-PBX Interface installation and data administration* (553-2811-200)
- *Digital Trunk Interface/Computer-to-PBX Interface maintenance* (553-2811-500)
- *Traffic measurement formats and output* (553-2001-450)
- *Spares planning guide* (553-2201-181)
- *Summary of transmission parameters* (553-2201-182)
- *X11 input/output guide* (553-3001-400)

Note: For general reference information on carrier systems and synchronization, see:

- Digital Network Notes, published by Telecom Canada, Ottawa, 1983
- AT&T Technical References 43861 and 41451
- EIA publication PN-1429

Note: Throughout this document, Meridian 1 refers to Meridian SL-1 machines and Meridian 1 system options, unless otherwise noted.

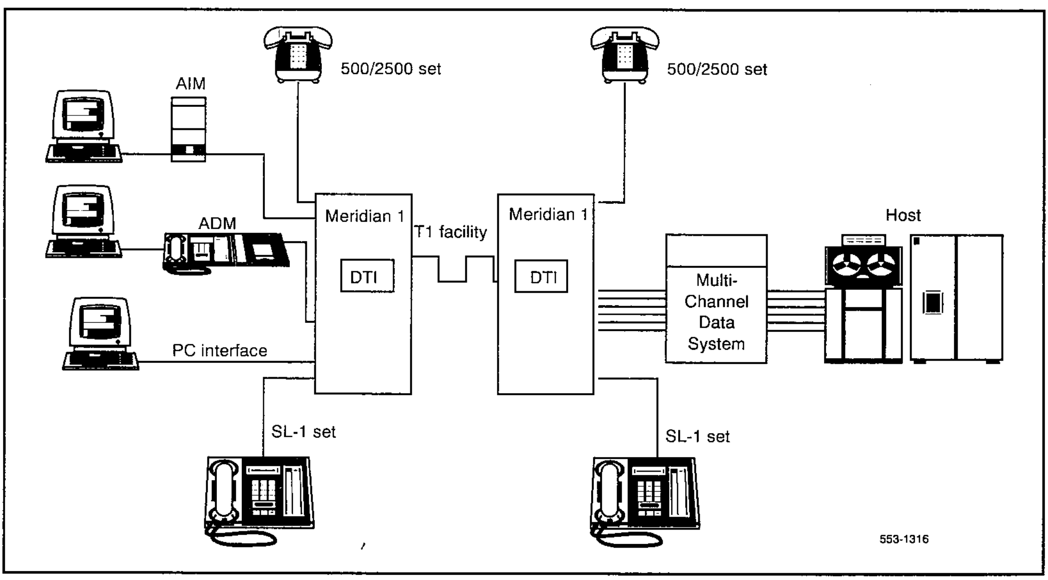
Digital Trunk Interface feature

The Digital Trunk Interface (DTI) provides the means for transmitting digital voice and data between a Network loop and a DS-1 digital carrier. DTI operates similarly to a channel bank for the carrier side and analog trunks for the Meridian 1 side. It integrates Meridian 1 digital switching capabilities and the most commonly available digital transmission facilities. It processes digitally both the transmission and reception of Meridian 1 data (mixed voice/ data), and voice calls.

DTI interfaces to DS-1/D3 digital carriers, which may use infrared, fiber optics, microwave radio, satellite link, or leased T1 facilities and may connect to:

- another Meridian 1 (see Figure 1) or SL-100
- a non-Meridian 1 system that can use T1 carrier facilities
- a digital Central Office (CO)

Figure 1
Typical Meridian 1 to Meridian 1 communication by means of DTI

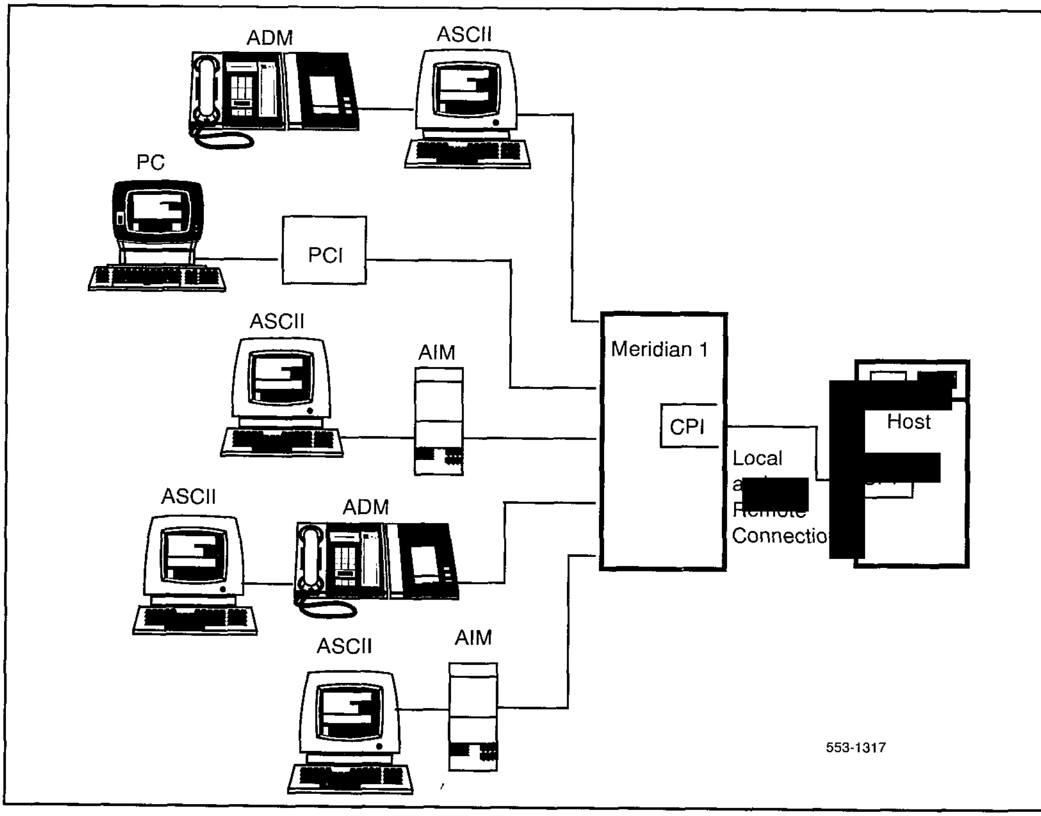


Computer/PBX Interface application

The Computer/PBX Interface (CPI) is a special application of the DTI feature that uses DTI to connect to a computer vendor-supplied interface residing on the vendor's host computer. See Figure 2.

The CPI application can be used without Clock Control on systems that support X11 software if the Meridian 1-to-Host connection is direct and not over an external network. External networks are applicable only to Meridian 1 systems and require network Clock Controllers.

Figure 2
Typical Meridian 1 to host computer communication by means of CPI



Benefits

DTI enhances Meridian 1 capabilities by:

- extending its capabilities to create all digital networks
- utilizing North American standard T1 facilities
- providing full compatibility with T1 digital protocol
- providing a single DTI to support the equivalent of 24 trunks
- eliminating need for channel banks
- eliminating need for modems on data transmission
- providing a single network that handles both voice and data
- configuring voice and data capacity on an "as-needed" basis
- substantially reducing the amount of equipment to interface 24 data channels to a host computer equipped with CPI

North American T1 carrier DS-1 frame organization

The North American T1 carrier operates at 1.544 Mbps using 193-bit frames in DS-1 format. Each frame contains an 8-bit data sample from each of 24 channels, plus 1 framing bit (F-bit), multiplexed into the 125 ms time period provided by the 8 KHz sampling rate. This establishes the basic rate of 1.544 Mbps. Signaling is done on a per-channel basis by periodically robbing Bit 8 from certain data samples (see Figure 3).

The F-bit is time-shared to identify both the channel and the signaling frame within a Superframe made up of 12 frames (see Figure 4). Terminal framing (FT) identifies the location of time slot 1, signaling framing (FS) identifies those frames in which two signaling bits, A and B, are transmitted on a time-shared basis. The least significant bit (Bit 8) in the sixth and twelfth frames are the signaling bits in each of the 24 channels.

Figure 3
DS-1 frame format

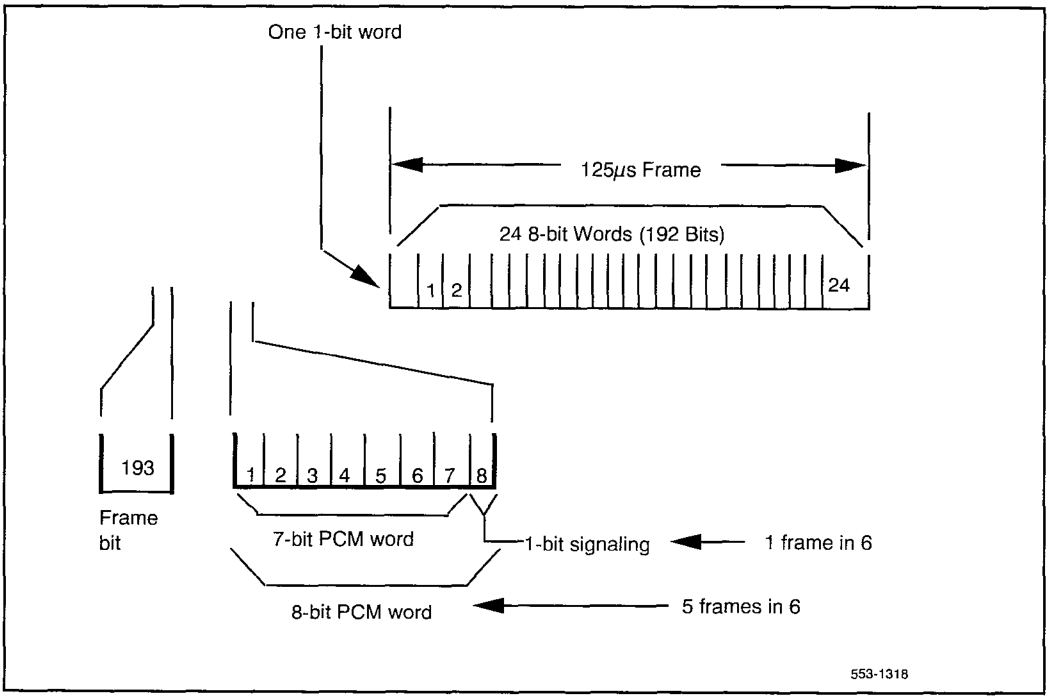
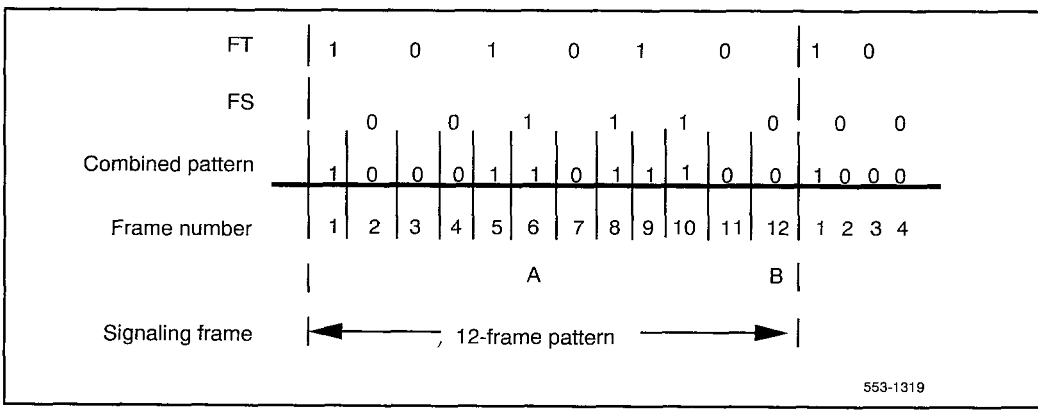


Figure 4
Framing format



B8ZS line coding method

With the increasing demand for higher bandwidths in data transmission, the use of the 8th bit of every channel in every frame of the DS-1 signal has proved to be expensive. The Binary 8 Zero Substitution (B8ZS) line coding method replaces a string of eight zeros with a standard pattern. This pattern has two intentionally added bipolar violations in the signal. At the receiving end, the detected pattern is replaced by the string of eight zeros.

This technique, coupled with the Extended Superframe format, increases the data rate from a maximum of 56K bits per second to the full 64 Kbps if in-band signalling is replaced with out-band signalling. A typical use is with ISDN Signalling Link, described in *ISDN Primary Rate Interface description and administration* (553-2901-100).

Extended Superframe format

The Extended Superframe (ESF) format consists of 24 frames. The 8 Kbps F-bit channel is divided into three separate channels:

Channel 1 Framing Pattern Sequence (FPS) - beginning with frame 4 or ESF bit 579, the framing bit of every fourth frame forms FPS 001011, which is used to determine the mainframe, superframe and robbed bit signaling synchronization. This sequence is a 2 Kbps channel.

Channel 2 Facility Data Link (FDL) - This is a 4 Kbps channel. The QPC720 uses FDL to convey remote alarm information or to transmit all ones, as selected in service change.

Channel 3 Cyclic Redundancy Check (CRC) - The CRC sequence is a 2 Kbps channel which carries the CRC-6 code. CRC indicates one or more bit errors in a block, or bits from the received bit stream. CRC can be used as an end-to-end bit error rate indicator.

The assignments of the F-bit and the A, B, C, and D bits are shown in Table 1.

Table 1
Extended Superframe Format (Part 1 of 2)

Frame number	F-bit				Robbed bit signaling
	Bit number	Assignments			
		FPS	FDL	CRC	
1	0	—	m	—	
2	193	—	—	CB1	
3	386	—	m	—	
4	579	0	—	—	
5	772	—	m	—	
6	965	—	—	CB2	A
7	1158	—	m	—	
8	1351	0	—	—	
9	1544	—	m	—	
10	1737	—	—	CB3	
11	1930	—	m	—	
12	2123	1	—	—	B
13	2316	—	m	—	
14	2509	—	—	CB4	
15	2702	—	m	—	
16	2895	0	—	—	
17	3088	—	m	—	
18	3281	—	—	CB5	C
19	3474	—	m	—	

Table 1
Extended Superframe Format (Part 2 of 2)

Frame number	F-bit				Robbed bit signaling
	Bit number	Assignments			
		FPS	FDL	CRC	
20	3667	1	—	—	
21	3860	—	m	—	
22	4053	—	—	CB6	
23	4246	—	m	—	
24	4439	1	—	—	D

Functions

Digital Trunk Interface

The QPC472/QPC720 Digital Trunk Interface (DTI) card is the interface between any selected channel of the 32 channel, 2.048 Mbps bit stream on the associated Meridian 1 Network loop, and a DS-1, 24 channel, 1.544 Mbps, bipolar carrier terminal.

DTI software

The DTI software module performs the following procedures in supporting the DTI feature:

— processes messages from the DTI hardware

— handles transmission loss pad settings

— handles Echo Canceller control

— processes on-line performance monitoring functions of the DTI

— inserts and translates call types (voice or data)

— converts a DTI TN to an equivalent Loop and Channel or Loop and Channel into an equivalent TN

— periodically checks the Clock Controller

— switches the reference clock from primary to secondary source if unable to track on primary

— switches the reference clock from secondary to free-run if unable to track on secondary

— updates the tracking of the Clock Controller after a change of primary or secondary reference source, or after SYSLOAD

Meridian 1/DS-1 conversion

Flexible mapping of 30 network time slots into 24 DS-1 channels (conversion from bit-interleaved to byte-interleaved format) and vice versa is performed by the DTI.

Idle, unequipped, or disabled DS-1 channel states

For each DS-1 channel that is idle or not configured, DTI hardware transmits "idle" (7F hexadecimal) or "unassigned" (FF hexadecimal) Pulse Code Modulation (PCM) codes respectively during the Voice/Data time slots. The DTI hardware and software data blocks are initialized to the appropriate idle or disabled state after system configuration or following service change and after every initialization. Other DS-1 systems receiving these signals interpret them as idle and unassigned, respectively.

Processing of A & B signaling bits

The DTI software sets the transmitted A & B bits to represent the appropriate signals for the trunk being supported. It also interprets the received A & B bit states as appropriate signaling states for that trunk.

Control of pad switching

The DTI inserts digital pads for both transmit and receive directions on a per channel basis, to achieve the desirable port to port transmission loss values. DTI channels are classmarked for different trunk types for pad setting and signaling purposes.

Meridian 1 data protocol converter

The Meridian 1 data protocol, which uses 8 bits for data, is not compatible with DS-1 protocol. Transmission on DS-1 facilities requires a "ones" density not compatible with data. Also, if A & B bits are processed on the link, the least significant bit of the 8 bit PCM word in the signaling frames is used for A & B signaling. For this reason, protocol conversion is provided to originating and terminating data messages on a per call basis at the originating and terminating DTIs. After trunk, line, and address signaling is complete and a path has been established between the two data terminals, the protocol converter is switched in until the call is disconnected.

Echo Canceller control

Echo Cancellers are required for satellite applications only. The DTI software controls external Echo Cancellers on a per channel basis by means of an RS232-C link. The Echo Canceller usually remains enabled. For Data calls however, the DTI hardware disables the Echo Canceller. When the Meridian 1 is connected to another switching node equipped with Echo Cancellers which do not provide per-call control, specific time slots can be permanently marked as data only.

Maintenance functions

The DTI performs maintenance functions related to the DS-1/D3 format link, including monitoring and reporting. It also does a self-check.

Data transmission and reception

DTI transmits and receives bipolar return-to-zero data at DS-1/D3 signal levels. Zero code suppression is normally in effect but may be inhibited as required by DTI software.

Controlled slips

DTI deletes or repeats frames of information from time to time in the receive PCM direction, to allow for signals which may be running at a slightly different bit rate to the DTI (that is, controlled by the transmitting system clock).

Clock Controller synchronization

Digital communication requires accurate timing alignment of digital signals so that data is interleaved into or extracted from the appropriate timeslots during multiplexing and demultiplexing operations. Frame synchronization can be attained by having all switches and transmission facilities in a network controlled by clocks of the same frequency. This is achieved by having all nodes (switches) in a network locked to a primary frequency source (a master clock).

Synchronization criteria

North American digital network nodes are synchronized using a priority master-slave method. Digital nodes are ranked in categories (AT&T Strata 1-4 or Bell Canada Node Category A to E). Each node is synchronized to the highest ranking node in its vicinity with which it has a direct digital link.

The Clock Controller meets AT&T Stratum 3 and Bell Canada Node Category D synchronization criteria. The clock has a drift rate of less than one part in 10^9 per day when the clock is in free running mode.

Clock control

The clock synchronization subsystem for DTI is provided by a single card on single CPU systems or a duplicated pair of Clock Controller (CC) cards on dual CPU systems. On dual CPU systems, only one Clock Controller is active at any given time. The disabled clock serves as a standby to the first.

When the clock subsystem is synchronized to an external clock source, through a specific DTI defined as a reference clock source, it is said to run in Tracking Mode. When no reference source is defined, or when the clock subsystem is not locked onto the external clock source, it is in Non-Tracking Mode and Running Free. In this case, there is no synchronization between the Meridian 1 and the external network. Where more than one DTI loop exists, Primary and Secondary Reference Clock sources can be defined for Tracking Mode operation.

The active CC in the subsystem performs the following functions:

- synchronizes to a Primary or Secondary Reference Clock (in Tracking Mode)
- supplies Meridian 1 with a clock

Tracking supervision

Meridian 1 software periodically (every 15 minutes) monitors the Clock Controller status. If the CC is unable to track the reference source, Meridian 1 sends a message to the maintenance TTY. If automatic switching is defined in the CC data block, the Meridian 1 switches as follows:

Unable to track on	Switch CC to
Primary	Secondary
Secondary	Free-run

Meridian 1 performs the same operation if a DTI that is a Primary or Secondary Reference Clock source is placed out of service.

Bit error rates

Bit error rate monitoring detects errors in transmission. There are two methods of bit error rate monitoring, bipolar violation tracking and cyclic redundancy check. The method used depends on the framing format (D2, D3, D4 or ESF). Framing format is defined in LD17 prompt DLOP.

Bipolar violation (BPV) In a bipolar pulse stream, pulses alternate in polarity. If, after transmission, two pulses of the same polarity are received in succession (this could be caused by an electrical disturbance, such as noise), a bipolar violation has occurred.

Cyclic redundancy check (CRC) The primary difference between BPV and CRC is that bipolar violations indicate errors on the local span, while CRC indicates errors on an end-to-end span. For example, on a satellite link, BPV only detects errors in the span between the Meridian 1 and the satellite connection. Since CRC travels between the entire span, it indicates end-to-end bit error rate. In DTI with ESF, bit error rate checking is done as CRC; however, the BPV counter is incremented.

Frame slip

Digital signals must have accurate clock synchronization for data to be entered into or taken from the appropriate timeslot during multiplexing and demultiplexing operations. Frame slip monitoring detects frame deletion and repetition errors in clock synchronization.

Frame alignment

Loss of frame alignment monitoring detects out-of-frame conditions on the DS-1 bit stream.

Loss of frame alignment thresholds DTI hardware detects out-of-frame conditions. The midnight routines print the number of loss of frame alignment occurrences and clear the counters.

There are three frame alignment thresholds set in LD73. When a maintenance or out of service threshold is reached, a DTA message is output as shown below.

- DTA019: Frame alignment maintenance limit.
- DTA020: Frame alignment out of service limit.

If a loss of frame alignment condition persists for three seconds, the affected DTI loop is taken out of service and a RED alarm is raised. If the loss of frame alignment condition clears for at least 15 seconds, the DTI is automatically restored to service. The following DTA message is generated:

- DTA021: Loss of frame alignment has persisted for 3 seconds.

Clock distribution

Multi-Group Mode (XN and XT) Each Clock Controller continuously drives the following signals to the IGS cards:

- M8 (8 MHz system clock)
- MFS (4 kHz sync pulse)
- CLKEN (clock enable active/disabled)

Single- or Half-Group Mode (N, NT, and RT) Each Clock Controller continuously drives the following signals to PS cards:

- M8XB (8 MHz system clock)
- MFSB(4 kHz sync pulse)

Half-Group Mode (MS, ST) The optional single Clock Controller operates in the half-group mode and supplies the clock. Should the clock fail, there is no switchover, and the PS card provides the clock.

Clock switchover

Each of the two Clock Controller cards monitors the other for clock failure and responds to switchover requests from its twin or from the CPU.

Phase locked loop

Clock Controller synchronization is achieved by phase locking to a 4 kHz clock derived from the 1.544 MHz reference source selected by the CPU. If the reference clock is lost, the CCs run free.

Automatic clock recovery

An option for automatic clock recovery can be selected in LD60 with the command EREF.

A DTI loop is disabled when it enters a red alarm (local) condition. If the red alarm is cleared, the loop is enabled automatically. When the loop is enabled, clock tracking is restored in the following conditions:

- If the loop is assigned as the primary reference clock but the Clock Controller is tracking on the secondary reference or in free run mode, it is restored to tracking on primary.
- If the loop is assigned as the secondary reference clock but the Clock Controller is in free run mode, it is restored to tracking on secondary.

If the 15-minute clock check indicates the system is in free run mode:

- Tracking is restored to the primary reference clock if defined.
- If the primary reference is disabled or in red alarm, (local) tracking is restored to the secondary reference clock if defined.

If the EREF option is selected in LD60, tracking on the primary or secondary reference clock is automatically switched in the following manner:

- If software is unable to track on the assigned primary reference clock, it switches to the secondary reference clock and sends appropriate DTC maintenance messages.
- If software is unable to track on the assigned secondary reference clock, it switches to free run.

Feature operation

DTI emulates both analog trunks and digital channel banks, while acting as an interface between Meridian 1 and digital transmission lines. DTI outputs digital signals at DS-1 level in either D2 or D3 format and with Release 11 or later software in D4 or ESF formats. The appropriate trunk signaling, basic trunk states, timing, and trunk processing are the same as the analog processes.

Interface functions are handled by DTI software:

— input/output messages

— pad setting

— Echo Canceller control

— data call recognition

— data call signaling to the next switch

These functions are under control of the trunk and DTI software modules. DTI supports both Dual Tone Multifrequency (DTMF) and dial pulse address signaling (the traffic provisioning ratio for DTMF is the same as provided for analog trunks).

Voice call processing

A general description of voice call processing is given in the following paragraphs.

Idle condition A digital trunk (DTI channel) must be idle to be available for outgoing calls. Idle state indications are different for different trunk types. These are handled automatically by DTI. When a trunk (channel) is idle, the DTI, under control of the CPU, sends idle PCM code (7F hexadecimal) on the Voice/Data time slot of the idle channel.

Outgoing calls

Trunk seizure When a caller dials a digital trunk route, the digit processor selects an idle trunk (a DTI channel) associated with the dialed route and sends seizure and pad control messages to the DTI hardware.

> *Note:* The route consists of digital trunks only. Analog and digital trunks are not mixed in a route, but a DTI loop can have 24 routes.

The DTI hardware sets the proper digital pads, using A & B bit signaling, sends an "off hook" message which is detected at the far end as a change of state. When it is ready to receive, the far end trunk (depending on its supervisory mode) informs the DTI by sending the appropriate A & B bit signal. After receiving this message, the DTI sends a seizure message to the far end. This is equivalent to an analog trunk closing the loop and must be done within 210 ms to avoid false signaling. The Echo Canceller, if equipped, remains enabled.

Outpulsing After receiving the seizure signal, or after an appropriate timing interval, the far end sends dial tone to the Meridian 1 and waits to receive DTMF or dial pulse address signaling, depending on the type of trunk (in the case of DTMF, digitized DTMF frequencies are transmitted on the active channels). The calling party dials the destination number. The Tone and Digit Switch (TDS) generates the pulses and timing on an enabled Network voice time slot. The DTI extracts this information and transmits DS-1 A & B bits to the far end in the appropriate signaling frame.

Answer When the far end receives the address signals, the call is routed to the dialed destination. If the called party answers, the far end returns an A & B bit answer signal to the DTI. DTI, in turn, notifies the CPU and a speech path is cut through.

Release Guard and disconnect supervision timing, and idling of the trunk are the same in operation as those of the analog trunk. After receiving a release message from the CPU, the DTI sends a release message (idle state) to the far end by means of A & B signaling bits. When the CPU sends the idle trunk message to the DTI, it starts sending the Idle PCM code on non-signaling frames on the idle channel.

Glare When a glare condition occurs (both ends of the same trunk attempt seizure at the same time), priority is given to the incoming call. In this case, the operation is the same as an analog trunk.

Incoming calls

Trunk seizure When the DTI hardware detects an A & B bit change of state on an idle DTI channel, it notifies the CPU. The CPU interprets the A & B bit state and takes the following action:

- **Central Office trunk** The CPU tells the DTI hardware to set up digital pads. It also alerts the Attendant or Night station DN (except on Direct Inward System Access (DISA) trunks).
- **TIE or Direct Inward Dialing Trunks** The CPU attaches the Digitone Receiver in the case of DTMF trunks, or interprets the A & B bits for the destination address in the case of dial pulse trunks. The address signals are processed by the digit processor and the called party is alerted. The CPU tells the DTI to set up digital pads for the call.

Answer When the call is answered, Meridian 1 software sends the appropriate answer message to the DTI. The DTI informs the far end and cuts through a speech path.

Release Guard and disconnect supervision timing, and idling of the trunk are the same in operation as those of an analog trunk. DTI relays the release message for the CPU (idle state) to the far end by means of A & B signaling bits. When the CPU sends the idle trunk message, the DTI starts sending the Idle PCM code on non-signaling frames on the idle channel.

Data call processing

Trunk timing and trunk transition used to set up data calls are similar to those used to set up voice calls with the following exceptions.

Mixed voice/data

When data calls using a route with mixed voice/data assignments are switched through a private network, intermediate switches must be signaled during call set up. This is done to ensure that:

– the tandem connection is made to digital facilities if Automated Modem Pooling is not provided

– the digital pads are not applied

– the Echo Cancellers are disabled on the data connections

This signaling is achieved by using the special address prefix or an equivalent format such as ESN "call type" network signaling. Signaling on DTI for STD and ESN trunk group option (shown in Table 2) is defined in LD16 in response to the SIGO (signaling arrangement) prompt.

Table 2
Network signaling on DTI

Trunk group	Call type	Digit	Sub-call type	Digit
Option (SIGO)				
STD	SIG STD VOICE	1		
	SIG STD DATA	2		
ESN5*	SIG DTI DATA	7	DTI CCBQ ALLOW #	1
			DTI STD NCOS #	2
			DTI STD TCOS +	3
			DTI SAT CCBQ #	4
			DTI SAT NCOS #	5
			DTI SAT TCOS +	6

* Call types 1-6 are treated as voice call. They have the same meanings as in ESN2 ESN3.

\# Followed by a 2-digit NCOS

\+ Followed by a 2-digit TCOS

Voice only

When voice calls are switched through a private network on a voice-only route, special ESN signaling is not required. Digital pads are applied as required, and Echo Cancellers are permanently enabled.

Data only

When data calls are switched through a private network on a data-only route, special ESN signaling is not required. Digital pads are not applied and Echo Cancellers are permanently enabled.

Data call limitations

Analog and digital calls may not be mixed in the same route. However, in cases where an analog facility runs parallel to a digital link, voice calls from the analog route are allowed to overflow onto the digital route and vice versa. Digital data calls do not overflow onto analog routes.

To prevent the inadvertent generation of false yellow alarms on a digital trunk loop, the number of data calls allowed on a digital trunk loop is limited to a value which is configured on a loop basis. This is done in LD17 in response to the DLOP (digital trunk loop) prompt. The range of possible values for maximum number of data calls is 0 to 24. The default value is 24. False yellow alarms can be caused by high-speed synchronous data being present on ***all*** channels for an extended period of time. If this special condition is a provisioning criterion, then consult your Northern Telecom sales engineer.

Recognition of data calls

Outgoing calls A bit in the terminal block identifies whether the call is from a voice or data terminal.

Incoming calls Incoming calls are identified by their special call type or route type (that is, dedicated for voice or data usage).

Digital connectivity

Digital data routes must be engineered so that end-to-end connectivity is provided. Intermediate switches must route data calls over digital trunks and must not insert pads or Echo Cancellers (call identity digits are not stripped unless the succeeding loop has dedicated voice or data channels).

Digital pad control

DTI software instructs its hardware to switch off digital pads for a particular channel when a digital data call is initiated for that channel. This is done to maintain bit integrity on such calls.

The DTI software inserts digital pads for both transmit and receive directions (on a per channel basis) to achieve the desirable port-to-port transmission loss values for each type of connection as identified by the Channel Classmarks.

> *Note:* The net loss of a particular overall port-to-port connection is the combination of this DTI pad loss *and* the pad loss of the connecting analog line. All of the various insertion loss specifications for DTI overall port-to-port losses are listed in *Summary of transmission parameters* (553-2201-182) and conform to the EIA document PN-1429.

DTI/CPI channel classmarks

Channel Classmarks (see Table 3) are used for different types of connections for signaling and pad-setting purposes. Digital pad values are assigned for the appropriate time slot (in accordance with the Pad Switching Table) during call processing when both the near and far end ports are known. When both ports of a connection are digital, only one port is assigned a pad value. The other port is automatically assumed to be zero.

Table 3
DTI channel classmarks

Number	Class	Explanation
1	ONS	Line interface/on premise line
2	OPS	Line interface/off premise line
3	ATT	Analog trunk interface/analog TIE trunk
4	AOT	Analog trunk interface/analog satellite PBX TIE trunk when PBX includes OPX telephones
5	SCTT	Digital trunk interface/combination satellite PBX TIE trunk
6	DTT	Digital trunk interface/digital or combination TIE trunk
7	ACO	Analog trunk interface/analog CO trunk
8	DCO	Digital trunk interface/digital or combination CO trunk
9	ATO	Analog trunk interface/analog toll office trunk
10	DTO	Digital trunk interface/digital or combination toll office trunk
11	SDTT	Digital trunk interface/digital satellite PBX TIE trunk
12	SATT	Analog trunk interface/analog satellite PBX TIE trunk

Echo Canceller

DTI hardware, under control of DTI software, disables the Echo Canceller on a per channel basis on data calls over satellite. If the Echo Canceller cannot be controlled dynamically (that is, on a per-call basis), some channels should be set to voice only and some to data only. Echo cancellation should then be manually removed for the data only channels.

Tandem switches

A Meridian 1, acting as a tandem switch, can perform the following tasks on data calls:

- recognize the special call type for data calls
- absorb the prefix if necessary
- route data calls to the appropriate trunks
- suppress digital pads
- disable Echo Cancellers
- if necessary, insert and send special call type prefix to the next switch down the line

In certain cases, one or more of these requirements may not be needed. This depends on the signaling and transmission requirements of the tandem.

Trunk supervision

Trunk supervision signaling is dependent on the trunk type being emulated by the DTI.

Computer/PBX Interface application

The Computer/PBX Interface (CPI) uses DTI as a multiplex interface between Meridian 1 and a host computer. The operation of DTI when used in a CPI application is the same as for other applications. However, the following considerations must be made:

Channel assignments The DTI channels associated with CPI are classmarked Data Only and must be configured in one of the following ways:

- one-way TIE trunks arranged for manual outgoing service
- two-way TIE trunks arranged for manual outgoing and incoming service
- two-way TIE trunks arranged for manual outgoing and wink start incoming service

Remote (yellow) alarms Remote (yellow) alarm recognition and transmission are disabled in the DTI and are not provided at the host computer interface.

Echo Cancellers Echo Cancellers must be disabled or removed.

Synchronization If the transmission facilities require the Meridian 1 to be synchronized with the digital network, then the Clock Controllers must be used as DTI applications.

Note: Clock Controllers are not required for the CPI application (except if the connecting facility is repeatered and requires synchronization).

Public Switched Data Service

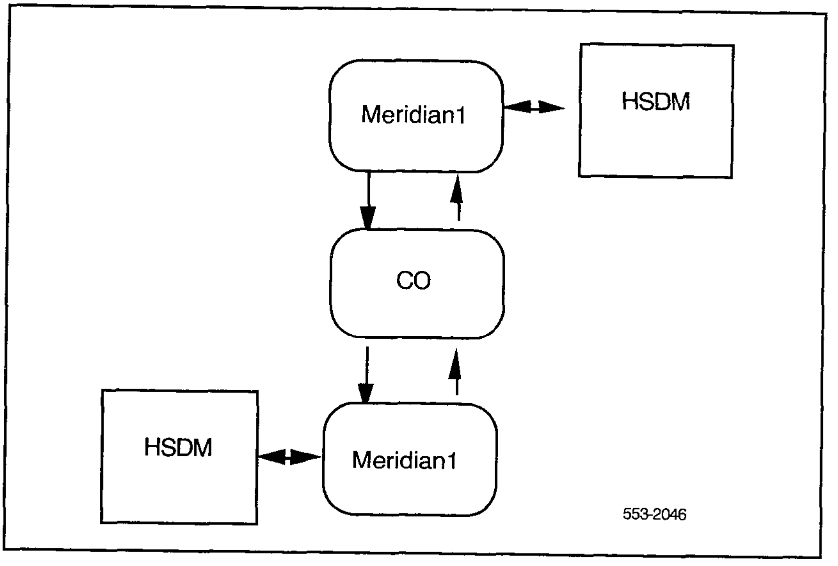
The Meridian 1 Public Switched Data Service is a digital data feature that provides a pure 56 Kbps data call between the following:

– a Meridian 1 and the Central Office (CO)

– a tandem call from an SL-100 to a Meridian 1

– a Meridian 1 and other Public Switched Data Service-compatible switches. See Figure 5.

Figure 5
Public Switched Data Service between Meridian 1 and Central Office



Note: For information on how to implement this feature, refer to *X11 features and services* (553-3001-305) and *X11 input/output guide* (553-3001-400).

The customer may install a T1 link to different vendors and use the Meridian 1 data unit to initiate or receive a 56 KBit digital data call. The digital data call then transports across the vendor's digital network to another Meridian 1 or SL-100.

The Public Switched Data Service supports DTI trunks, TIE and DID/DOD trunks, and Electronic TIE Network-compatible signaling (ETN).

Operation

The data selection (DSEL) in the route data block can be defined as voice calls only (VCE), data calls only (DTA), or voice or data calls (VOD). The call can be defined as voice calls, regular data calls, or Public Switched Data Service calls. Refer to *X11 input/output guide* (553-3001-400) to configure the route data block.

For direct access, the user dials the regular 7 or 10 digits DN. The translation selects the TIE or DID/DOD DTI trunks. If the normal public access trunks are analog trunks, then the data DN needs to be separated into an ESN special number so the user can access the digital DTI facility.

For special route access, the user dials a route access code after hearing a dial tone. All the trunks in that route are one of TIE or DID/DOD DTI trunks.

Configurations

End to End DTI network For all Meridian 1 networks (Point to Point), users can access the existing data facility in the Meridian 1 to support data calls or they can select the Switched 56 data mode. For mixed vendors private network, users can only select the Public Switched Data Service mode.

Receiving data calls

A call can either be answered automatically (by a switch located inside the data module) or manually (by pressing the data DN key on the data module)

Public Switched Data Service call on Primary Rate Interface The following routes are possible using this feature on Primary Rate Interface (PRI):

- **Point to point access** For point to point access of TIE trunks, the software can be modified to handle the requirements of this feature.
- **Tandem call** For tandem access, additional information on this feature is needed, or the data call can be defined as a voice call.
- **DID/FEX/WATS/Accunet** The Meridian 1 supports Public Switched Data Service data calls to these trunk types.
- **Public Network Hopoff** Signaling informs the tandem switch about the Public Switched Data Service data call.

Equipment

Required firmware

To implement the B8ZS feature requires CPU ROM firmware that is compatible with X11 release 19 software.

Installation-dependent hardware

The following hardware may be required, depending upon the specific installation:

- Echo Canceller

  A Tellabs 24 Channel Echo Canceller, or the equivalent, should be used for echo cancellation on DTI voice calls over satellite only.

- Channel Service Unit

  Depending on local or national regulations (for example, Part 68 of FCC regulations), connections to registered common carrier trunks may require that Network Channel Terminating Equipment (NCTE) be provided at the demarcation point. Digilink, Verilink, or similar Channel Service Units (CSUs) can be used.

- Test Equipment

  A Thor Error Counter, or equivalent, should be used for testing bipolar violations on DTI circuits.

Required hardware

To implement the DTI feature, Meridian 1 uses:

- QPC472 Digital Trunk Interface or QPC720 Primary Rate Interface (PRI)

  The DTI card or PRI card is associated with a Meridian 1 Network loop and forms the interface between the loop and a DS-1 24-channel 1.544 Mbps bipolar carrier terminal.

  Note 1: DTI loops that use the B8ZS feature must be QPC720 packs or equivalents.

  Note 2: This features applies to the 1.544Mbps DTI only. DTI2/PRI2 is not supported.

  DTI card location Each DTI card occupies two adjacent slots of a CPU shelf or a Network shelf. The specific location depends on the:

  - machine type
  - mode of operation
  - availability of space

  As many as six DTI cards can be plugged into a spare Network shelf along with a Power Converter card.

- QPC471 or QPC775 Clock Controller (CC)

  Clock Controllers are required when the Meridian 1 is synchronized to an external source clock, such as AT&T Stratum 3 (all DTI applications and CPI applications where the host and CPI are remote and require a repeatered facility). The CC generates and distributes clock signals to the system.

  Note 1: The QPC775 Clock Controller card is currently only available in Canada and the international market.

  Note 2: The QPC471 and QPC775 Clock Controllers cannot be intermixed in one system.

  Note 3: For the CPI application, in most cases a Clock Controller is not required.

  Note 4: Option 81 systems require QPC471 Vintage H or later.

QPC472 and QPC720 hardware description

Physical description

The QPC472 and QPC720 cards consist of two PCBs arranged into a single assembly in a mother/daughter configuration. The Network Interface (NI) PCB serves as the motherboard. It connects through a 60-pin ribbon cable to the Carrier Interface (CI) PCB daughterboard.

Note 1: QPC472 card's daughterboard is the QPC473.

Note 2: QPC720 card's daughterboard is the QPC752.

The DTI card draws power through the backplane connector of the NI. NI faceplate mounted connectors J1 and J2 extend through the faceplate and connect by cable to the Clock Controller when the DTI card is used as a reference clock source. J3 connects by cable to the Network loop.

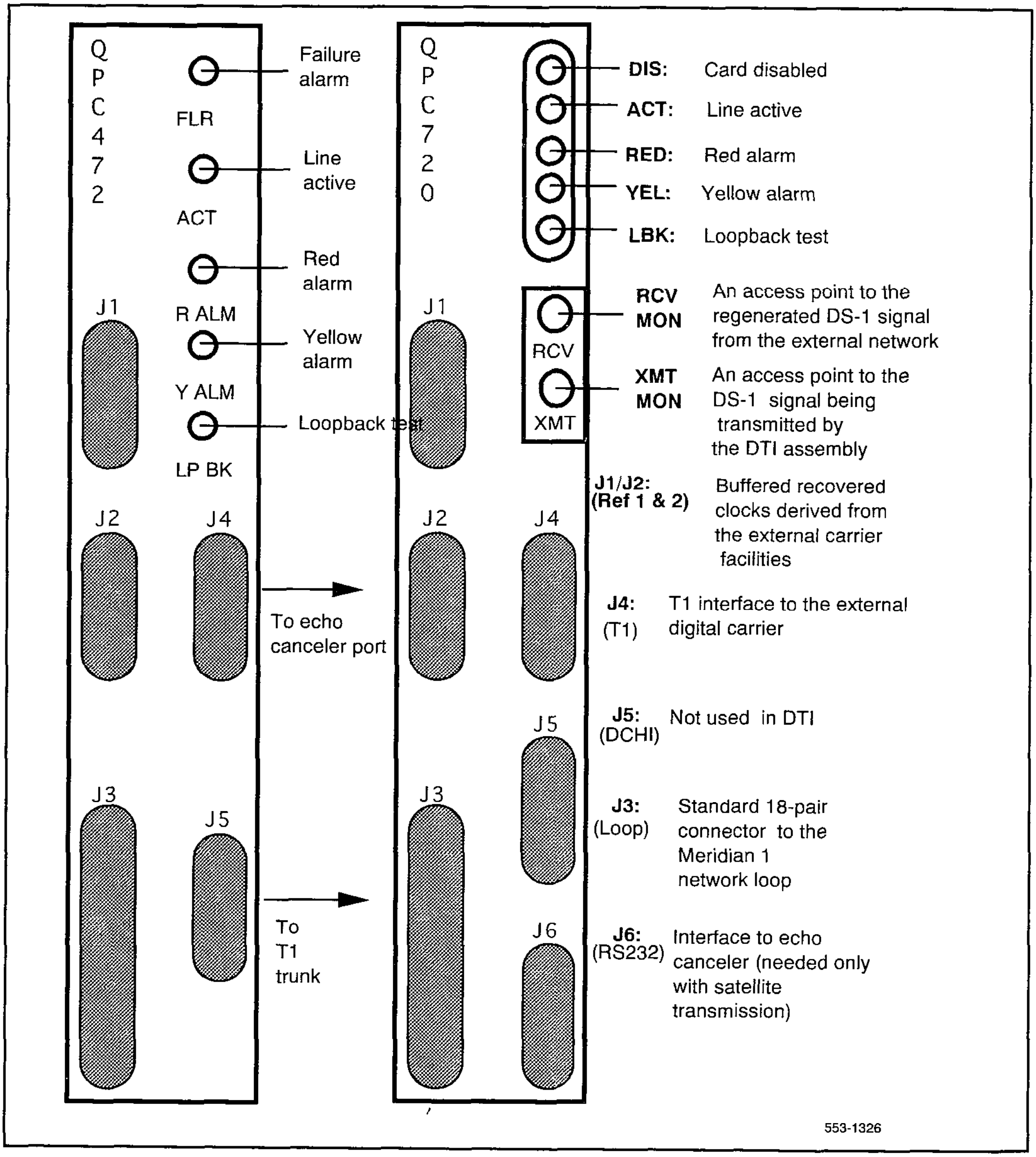
Because of its power requirements, a QPC472 or QPC720 card must be plugged into a network or CPU backplane.

QPC472 J4 and J5 Extending through the faceplate from the daughterboard are connectors J4 and J5. J4 connects by cable to the Echo Canceller. J5 connects by cable to the T1 trunk.

See Figure 6 for card faceplate jacks and LED locations.

QPC720 J4, J5, and J6 Extending through the faceplate from the daughterboard are connectors J4, J5, and J6. J4 connects by cable to the T1 trunk. J5 is not used in DTI/CPI applications. J6 connects by cable to the Echo Canceller.

Figure 6
QPC472/QPC720 faceplate jacks and LEDs



Faceplate LEDs

QPC472 The LEDs on the QPC472 faceplate indicate the following states:

FLR	DTI failed or is disabled if ON
ACT	DTI active if ON
RALM	DTI local (red) alarm
YALM	DTI remote (yellow) alarm
LP BK	DTI in loop-back test state if ON

QPC720 The LEDs on the QPC720 faceplate indicate the following states:

DIS	DTI failed or is disabled if ON
ACT	DTI active if ON
RED	DTI local (red) alarm
YEL	DTI remote (yellow) alarm
LBK	DTI in loop-back test state if ON

Transmission equalizer switch

Making appropriate DIP switch settings on the CI (daughterboard) card of the DTI card achieves transmission equalization. Transmission equalization is needed to compensate for cable-induced amplitude and phase distortions.

QPC472 Card The DIP switch is located on the QPC473 daughterboard, and is labeled SW1. The switch consists of 8 poles. Poles 1 through 7 are used for DTI applications.

QPC720 Card The DIP switch is located on the QPC752 daughterboard and is labeled SW2. The switch consists of 7 poles. All 7 poles are used for DTI applications.

See Figure 7 and refer to Tables 4 and 5 for DIP switch locations and settings.

Table 4
QPC472 transmission equalization switch settings for DTI

Switch setting (transmission equalization)	To repeatered facility	To cross-connect point
5 on	0 - 45 m (o - 150 ft)	0 - 15 m (0 - 55 ft)
2, 4, 6, on	45 - 135 m (150 -450 ft)	15 - 100 m (56 - 355 ft)
1, 3, 7, on	135 - 225 m (450 - 750 ft)	100 - 200 m (355 - 655 ft)
Note: All switch positions should be OFF except those under the 'switch settings' column.		

Table 5
QPC720 transmission equalization SW2 switch settings for PRI/DTI

Switch S2 settings	To repeatered facility	To cross-connect point
5 on	0-45 m (0 - 150 ft)	0 - 30 m (0 - 100 ft)
2, 4, 6 on	46 - 135 m (151 - 450 ft)	31 - 100 m (101 - 355 ft)
1, 3, 7 on	136 - 225 m (451 - 750 ft)	101 - 200 m (356 - 655 ft)

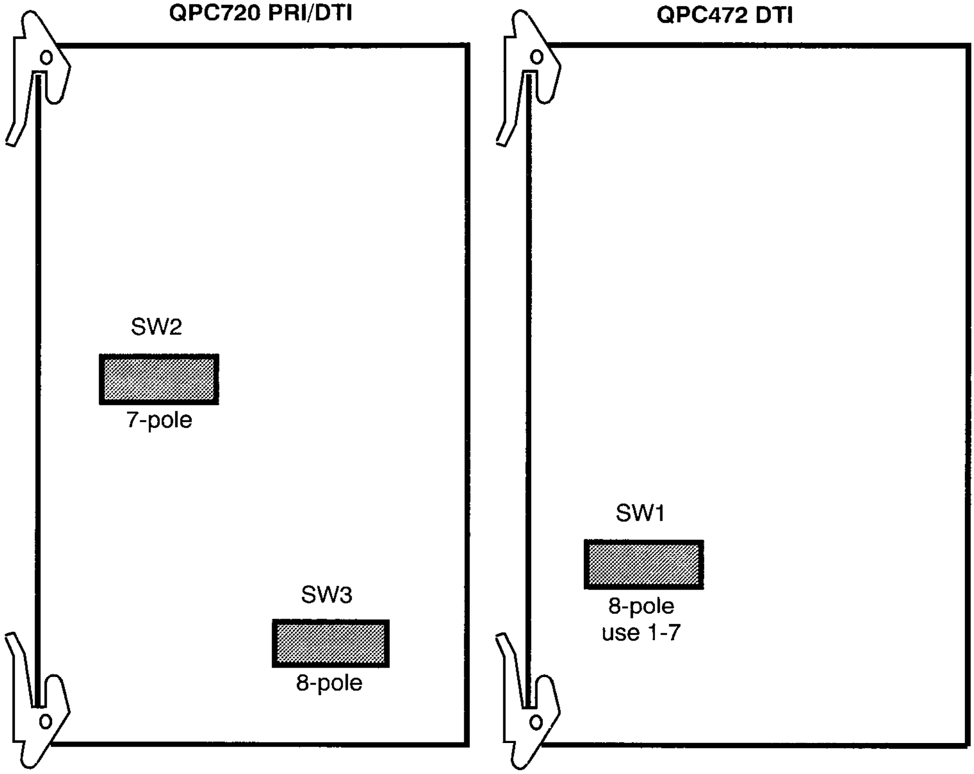
Switch 3 options for PRI/DTI with ESF

SW3-1	on	=	extended superframe format (ESF)
SW3-2	off	=	superframe format (SF)
	on	=	B8ZS line encoding
	off	=	AM1 line encoding
SW3-3	on	=	facility data link (FDL) yellow alarm method
	off	=	Digit 2 yellow alarm method

Note 1: All positions on S2 (location B22) are OFF except as shown under the column labeled 'Switch S2 settings.' The 8-pole SW3 (location E37) positions are OFF except as shown for 'Switch 3 options for PRI/DTI with ESF.'

Note 2: For D2, D3, or D4 framing formats (superframe formats), set all SW3 options to OFF.

Figure 7
QPC472/QPC720 DIP switch locations and settings



Note: See tables on PRI/DTI transmission equalization switch settings.

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Faceplate connectors

J1, J2 Reference Clock The DTI card is equipped with two 9-pin PCB mounted connectors designated J1 and J2. If the DTI is defined as the primary reference clock source, DTI connector J1 is cabled to CC0 connector J2 and DTI connector J2 is cabled to CC1 connector J2. If the DTI is defined as the secondary reference clock source, DTI connector J1 is cabled to CC0 connector J1 and DTI connector J2 is cable to CC1 connector J1.

J3 Network Loop The 36-pin DTI connector J3 is cabled to the 36-pin connector (even or odd) of its assigned Network loop.

QPC472 J4/QPC720 J6 Echo Canceller The DTI/Echo Canceller controller interface is made through the 15-pin female connector. The DTI is a DTE RS-232 interface. Its pin assignments and interconnections are shown in Table 6.

Table 6
DTI/Echo Canceller RS232-C interface pin assignments

Signal	EIA RS232-C circuit designation	Echo Canceller pin	DTI pin
Transmitted Data (TXD)	BA	5	2
Received Data (RXD)	BB	2	3
Request to Send (RTS)	CC	—	4
Clear to Send (CTS)	CB	—	5
Signal Ground (common return)	AB	10	7
Received Line Signal Detector (DCD)	CF	1	8
Data Terminal Ready (DTR)	CD	4	20

QPC472 J5/QPC720 J4 Carrier Interface The DTI interface to the carrier, by means of the Echo Canceller (if required), is made through the 15-pin connector. The pinouts at the DTI end are shown in Table 7.

Table 7
DTI /QPC472 J5/QPC720 J4 Carrier Interface pinouts

Pin	Signal	Pin	Signal
1	XTIP	3	RTIP
9	XRING	11	RRING
2	GRD	4	GRD

Note: See *Digital Trunk Interface/Computer-to-PBX Interface installation and data administration* (553-2811-200) for DTI cabling details.

QPC471 and QPC775 Clock Controllers

The QPC471 and QPC775 Clock Controller (CC) cards provide Stratum 3 clocking to an external facility and can therefore control synchronization. It can also synchronize the Meridian 1 Network to an external source clock and generate and distribute a clock signal to the system.

The Meridian 1 accepts several vintages of the QPC471. The QPC471H is required for option 81.

The QPC775A (currently available in Canada only) Clock Controller card is compatible with the N, NT, XN, XT, MS, ST and RT machine types.

Note: The QPC471 and QPC775 cannot be intermixed in one system.

The CC is contained on a 12.5 in. (32 cm) by 10 in. (25 cm) printed card. The CC connects with other circuits through a 120-pin backplane connector (P1) and through two 9-pin PCB-mounted connectors, J1 and J2. J1 and J2 extend through the faceplate and through a 50-pin faceplate mounted connector J3. An Enable/Disable switch and LED are also located on the CC faceplate.

Clock Controller Enable/Disable switch and indicator

When the Enable/Disable switch has been used to disable the Clock Controller, the LED is turned on and the Clock Controller does not distribute clock signals. Clock Controller Enable and Disable are also under both CPU control and that of the DTI hardware diagnostic, LD60.

Clock Controller card location Clock Controller cards are located as follows:

- XN - CPU shelf
- XT - CPU/MEM shelf slot 14 or 15
- N, RT - Network shelf slot 13
- MS - CE shelf slot 9
- ST, STE - CE shelf slots 5 through 8 or 10 through 12
- NT - Network shelf 0 (half group) slot 13; network shelf 1 (single group) slot 2
- 21, 21E - CE/PE Network slots 4 through 7
- 51 - CPU/Network slot 9

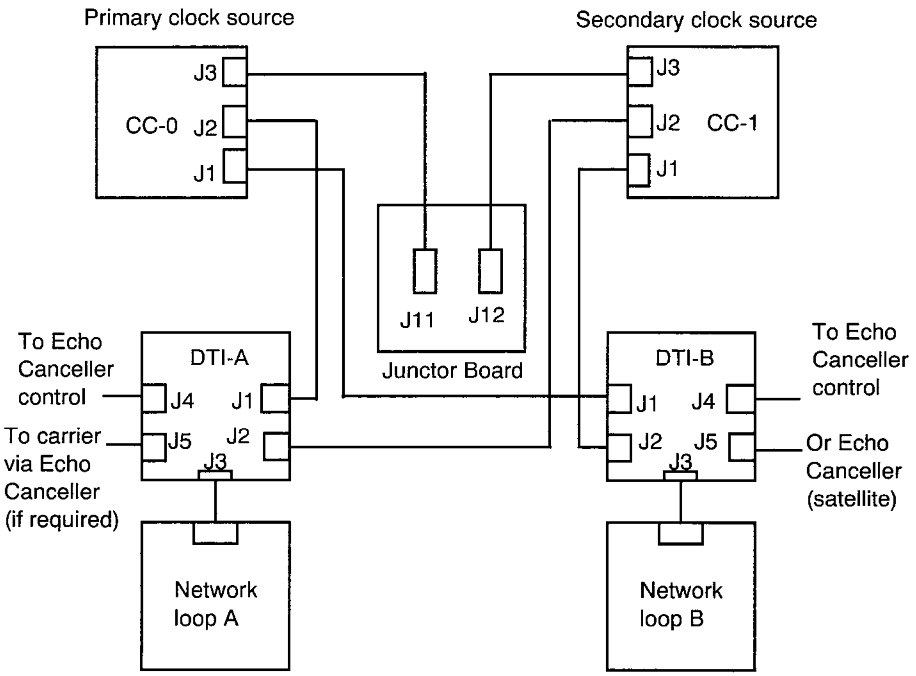
For dual-CPU systems, two Clock Controller cards are required (one for each CPU), in the following locations:

- 61 - CPU/Network slot 9
- 71 - CPU slot 14
- 81 - Core slot 6 (requires QPC471 vintage H)

Depending on the configuration of the Meridian 1, DTI and CC cards are arranged differently. Note that only the first two DTI cards are connected to the Clock Controllers. All other DTI cards are connected only to Network cards. This is because the Meridian 1 either slaves to the carrier clocking by means of the Stratum 3 level Clock Controller or provides clock to the carrier.

In dual CPU systems, Clock Controller redundancy is provided for each CPU. Northern Telecom recommends that dual-CPU systems operate with Clock Controllers of the same vintage. In addition, any two, but only two, DTI cards are connected to the CC cards for further backup. If one T1 link fails, the second DTI card can provide clocking to the CC cards. Any additional DTI cards receive clocking by means of the Network Signaling bus. The following figures depict some typical arrangements.

Figure 8
XN and XT multi-group operation



Note: This illustration shows how the QPC472 DTI card is connected. The following are the jack equivalents on the QPC720 DTI card:

QPC472 J4 = QPC720 J6

QPC472 J5 = QPC720 J4

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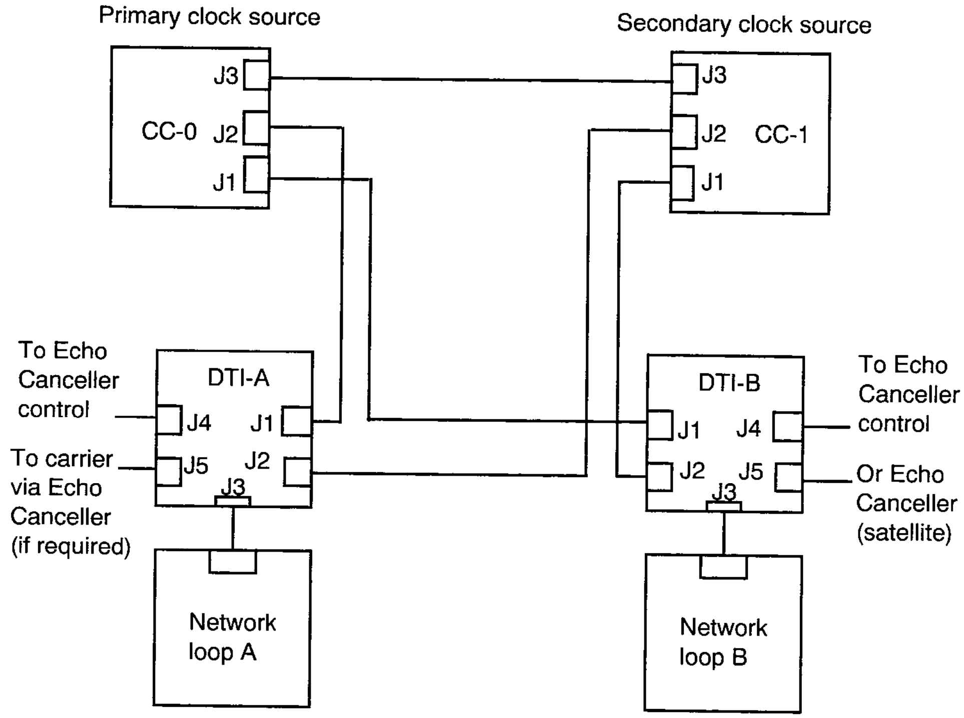
Figure 9
N and NT single-group operation



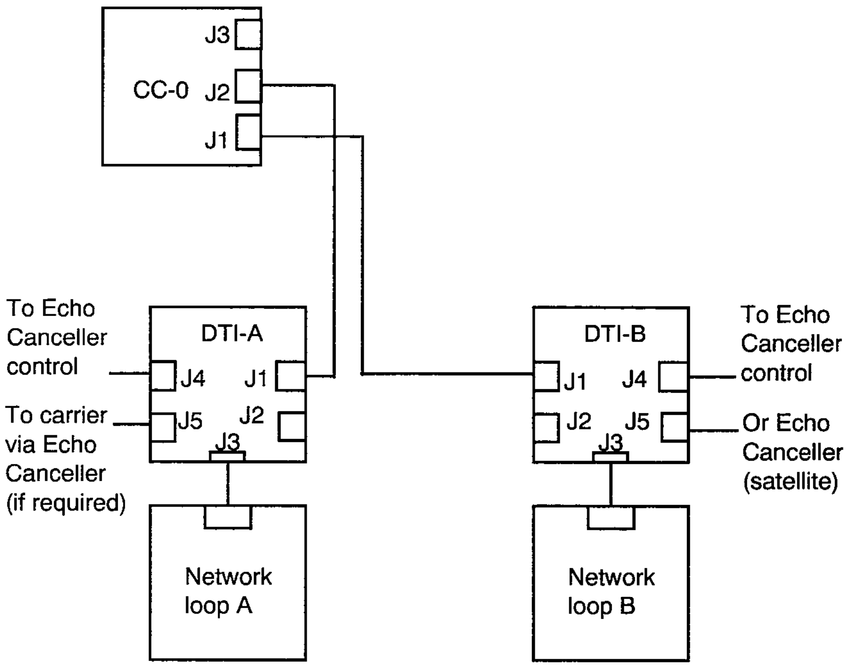
Note: This illustration shows how the QPC472 DTI card is connected. The following are the jack equivalents on the QPC720 DTI card:

QPC472 J4 = QPC720 J6

QPC472 J5 = QPC720 J4

553-1324

Figure 10
N, NT, MS, ST, and RT half-group operation



Note: This illustration shows how the QPC472 DTI card is connected. The following are the jack equivalents on the QPC720 DTI card:

QPC472 J4 = QPC720 J6

QPC472 J5 = QPC720 J4

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Clock Controller switch settings

Switch settings for the Clock Controller are shown in the following tables.

Table 8
QPC471 vintages A through G clock controller switch settings

Vintage	Switch	MS	ST, STE, 21A, 21	N, NT, RT, 51, 61	XN, XT, 71, 81
QPC471A	SW2	not applicable	not applicable	ON	OFF
QPC471B through G	SW1 SW2 Jumper F38 Jumper G38	ON ON TP9-TP10 TP12-TP13	ON OFF TP8-TP9 TP11-TP12	ON OFF TP8-TP9 TP11-TP12	OFF OFF TP8-TP9 TP11-TP12
QPC775	SW2	ON	ON	ON	OFF
	SW3	ON	OFF	OFF	OFF
	SW4(1,2)	OFF	ON	ON	ON

Note 1: ON means 1 or CLOSED

Note 2: OFF means 0 or OPEN.

Note 3: The QPC775 Clock Controller card is currently available in Canada only.

Table 9
QPC471 vintage H clock controller switch settings

System		SW1				SW2				SW4			
		1	2	3	4	1	2	3	4	1	2	3	4
ST, STE, 21A, 21, 21E		on	on	on	on	off	off	off	off	off	off	off	off
MS, SN		on	on	on	on	on	on	on	on	off	off	off	off
RT, N, NT, 51, 61		on	on	on	on	off	off	off	off	off	on	*	*
XN, XT, 71, 81		off	off	off	off	off	off	off	off	off	on	*	*
Cable length between the J3 faceplace connectors:													
0 - 4.3 m	(0 - 14 ft)									off	off		
4.6 - 6.1 m	(15 - 20 ft)									off	on		
6.4 - 10.1 m	(21 - 33 ft)									on	off		
10.4 - 15.2 m	(34 - 50 ft)									on	on		

* If there is only one clock controller card in the system, set to OFF. If there are two clock controller cards, set to match the cable length between the J3 faceplate connectors. Determine the total cable length (no single cable can exceed 25 ft) between the J3 connectors. Both cards must have the same setting.

Clocks per system

The number of clocks used with the DTI is dependent on the Meridian 1 Network Group configuration. Table 10 shows clocks required per system.

Table 10
Meridian 1 system clock requirements

Number of CC	Network Group	Meridian 1 type
2 (QPC471)	Multi-Group	XN, XT, 71
2 (QPC471)	Single-Group	N, NT, 61
1 (QPC471)	Half-Group	N, NT, RT, 51
1 (QPC471)	Half-Group	MS, ST, STE, 21, 21E
2 (QPC471H)	Multi-Group	81

Clock Controller interconnections

Backplane interface The CC interconnects with the CPU bus by means of the 120-pin backplane connector (P1).

Faceplate connectors The CC card is equipped with two 9-pin PCB-mounted connectors, designated J1 and J2. Connector J1 is associated with the DTI unit which is selected as the secondary reference clock source. J2 is associated with the DTI unit that is selected as the primary reference clock source. The CC is also equipped with a 50-pin, faceplate mounted connector, designated J3. It is connected to the junctor board in Multi-Group mode, or directly to the other CC when Single-Group mode is used.

> *Note:* CC cabling details are described in *Digital Trunk Interface/ Computer-to-PBX Interface installation and data administration* (553-2811-200).

Interfacing with common carriers

The DTI provides an interface to the DS-1/D3 line either directly or by means of channel banks or office repeater. In accordance with FCC regulations (FCC 68), connections to registered common carrier trunks require that Network Channel Terminating Equipment (NCTE) – such as the NT QRY551 Channel Service Unit – be provided at the demarcation point, as long as the facility uses cable (microwave, fiber, infrared, and direct satellite connections do not require NCTE equipment).

Echo Canceller interface

The DTI provides both a T1 line interface and a control interface to a remotely controlled Echo Canceller. The control interface is a serial link with a signal format compatible with EIA standard RS232-C. Both the DTI and the Echo Canceller act as a DTE. Installation of the Echo Canceller is covered in the manufacturer's documentation; however, the following guidelines outline the requirements for operation with DTI.

Physical interface The DTI/Echo Canceller interface is made through a 15-pin D-type connector at the DTI end. See Table 11.

Table 11
DTI/Echo Canceller interface

Interface protocol	Serial link protocol
Mode	asynchronous
Data transfer rate	4800 baud
Duplex	full
Received characters echoed	yes
Parity	even
Start bits	1
Data bits	7
Stop bits	1

Command protocol All remote control commands consist of a sequence of four 7-bit ASCII characters as shown in Table 12.

Table 12
Command protocol

Character	Description
1	System Unit Number (1)
2 & 3	Channel number being controlled (1-24)
4	Command Key (see Table 13)

There are two types of commands:

– **Control** causes reconfiguration of the specified channel. The Echo Canceller must return an ACK (06 Hex) if the command is legal or a NACK (15 Hex) if the command is illegal.

– **Status** monitors the existing configuration. The Echo Canceller must return a three-character response, the first two characters of which are the channel number. The third character is P (pass) or F (fail).

Table 13
Echo Canceller command keys

Command type	Command	On		Off	
		ASCII	Hex	ASCII	Hex
Control	bypass (Notes 1 & 3)	B	42	A	41
	off-hook (Notes 2 & 3)	N	4E	O	4F
	canceller only (Note 4)	C	43	D	44
Status	self-test result (Notes 1 & 2)	T	54	—	—

Note 1: Echo cancellation is disabled when input is ON. Echo cancellation is active when the device performs a self-test.

Note 2: Continuous echo cancellation takes place and self-testing is inhibited when input is ON.

Note 3: When Bypass and Off-Hook are entered simultaneously, the channel is removed from echo cancellation service.

Note 4: Residual suppressor is disabled but cancellation circuitry remains active.

SL-1

Digital Trunk Interface/Computer-to-PBX Interface

Description

Release 3.0
Standard
August 1, 1993
Printed in the U.S.A.

northern telecom

SL-1

Digital Trunk Interface/ Computer-to-PBX Interface

Installation and data administration

Publication number: 553-2811-200
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Document release: 5.0
Document status: Standard
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Revision history

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Standard, release 1.0. Reissued for compliance with Northern Telecom standard 164.0.

This document has been merged with *DTI/CPI wire list* (553-2811-200 Appendix 1). The information previously contained in that document is now contained in the chapter,"Wire list", in this document.

December 1, 1991

This document is reissued to include technical content updates. Due to the extent of changes revision bars are omitted.

December 31, 1992

This document is issued to include technical updates. Changes are noted with revision bars in the margins.

August 1, 1993

This document is reissued for updates and changes resulting from X11 release 19. All updates are noted with revision bars in the margins.

October 31, 1993

This document has been reorganized to enhance ease of use and updated to reflect current technical information. Technical changes are noted with revision bars in the margins.

Contents

List of figures

List of tables

Introduction

This document provides installation, data administration, and acceptance test procedures for the Digital Trunk Interface and Computer to PBX Interface (DTI/CPI) features.

These features are available on ST, STE, RT, MS, N, NT, XN, and XT systems and on Meridian 1 system options 21, 21E, 51, 61, 71, and 81. The DTI/CPI features are offered on the following X11 software releases:

— Release 5 and later software for N and XN systems

— Release 8 and later for M, NT, and XT systems

— Release 9 and later for ST systems

— Release 12 and later for RT systems

— Release 15 and later for system options 21, 21A, 51, 61, and 71.

— Release 18 and later for system options 21E, STE, and 81.

Note: Throughout this document, Meridian 1 refers to all Meridian systems unless otherwise noted.

The document includes the following sections:

— **DTI installation:** over view of DTI hardware, interfaces, and cable requirements, including installation and removal procedures for DTI cards, Echo Cancellers, and Clock Controllers.

— **Data administration:** procedures for provisioning the DTI feature and the CPI feature.

— **Acceptance tests**: acceptance test procedures for the DTI feature.

— **Wire list**: information on building external cables of nonstandard lengths for DTI interface applications.

Other documentation

Refer to the following Northern Telecom Publications for more information:

- *System installation* (553-3001-210)
- *Circuit card installation and testing* (553-3001-211)
- *ISDN Primary Rate Interface installation* (553-2901-200)
- *Digital Trunk Interface/Computer-to-PBX Interface maintenance* (553-2811-500)
- *X11 input/output guide* (553-3001-400)
- *Digital Trunk Interface/Computer-to-PBX Interface description* (553-2811-100)

DTI installation

This chapter describes the procedures for installing and removing DTI/CPI equipment and Clock Controllers in a variety of systems. The descriptions include cable requirements.

The first part of the chapter provides detailed information on the DTI card. A section on the Echo Canceller follows the DTI information. The last part of the chapter focuses on Clock Controllers.

Equipment overview

The DTI connects a Network loop to an external DS-1 interface for both voice and data transmission over 24 digital channels.

The DTI interface requires a **QPC472** DTI or **QPC720** Primary Rate Interface (PRI) card. Because this document deals with DTI functions, both cards are referred to as the "DTI" card or "DTI" hardware whenever their functionality is identical.

Clock Controllers synchronize the system clock rate with the clock rates of digital switching and transmission equipment over a T1 circuit. Dual CPU systems have two Clock Controllers: CC0 (associated with CPU 0) and CC1 (associated with CPU 1).

The standard Meridian 1 Clock Controller is the **QPC471**, currently available in vintages C, D, E, F, G, and H. Option 81 must use vintage H.

Customers outside the United States can use the **QPC775**. Vintage A of the QPC775 can be used with non-networked systems (systems linked directly to a central office); vintage B is appropriate for networked systems. (Meridian 1 to Meridian 1 configuration). Customers with option 81 systems must use QPC775 vintage C.

Note: A system cannot use both the QPC471 and the QPC775. A system can mix Clock Controller vintages A through G. Vintage H Clock Controllers can be used only with another vintage H Clock Controller.

DTI interfaces

The DTI card provides a connection to the DS-1 channel interface directly through an office repeater, or through an Echo Canceller.

In the U.S.A., FCC Part 68 regulations require Network Channel Terminating Equipment (for example, the NTQRY551 Channel Service Unit) installed at each connection point of a private system to a registered common carrier trunk.

Echo Cancellers are only required if satellite transmission is used. The DTI card provides both a T1 line interface and a control interface to link with a signal format compatible with EIA standard RS-232-C. Both the DTI and the Echo Canceller act as Data Terminal Equipment (DTE). The Echo Canceller's control protocol must conform with that of the Tellabs Model 251.

QPC472 DTI specifications

The QPC472 DTI only requires power and ground from the backplane. Power requirements are:

+5 volts at 3 amperes

+12 volts at 50 milliamperes

-12 volts at 50 milliamperes

The QPC472 DTI contains five LEDs and five external connectors. Table 1 shows the QPC472 transmission equalization SW2 switch settings for DTI for all systems. Table 2 lists the external connectors located on the QPC472 DTI faceplate. Figure 1 illustrates the faceplate layout.

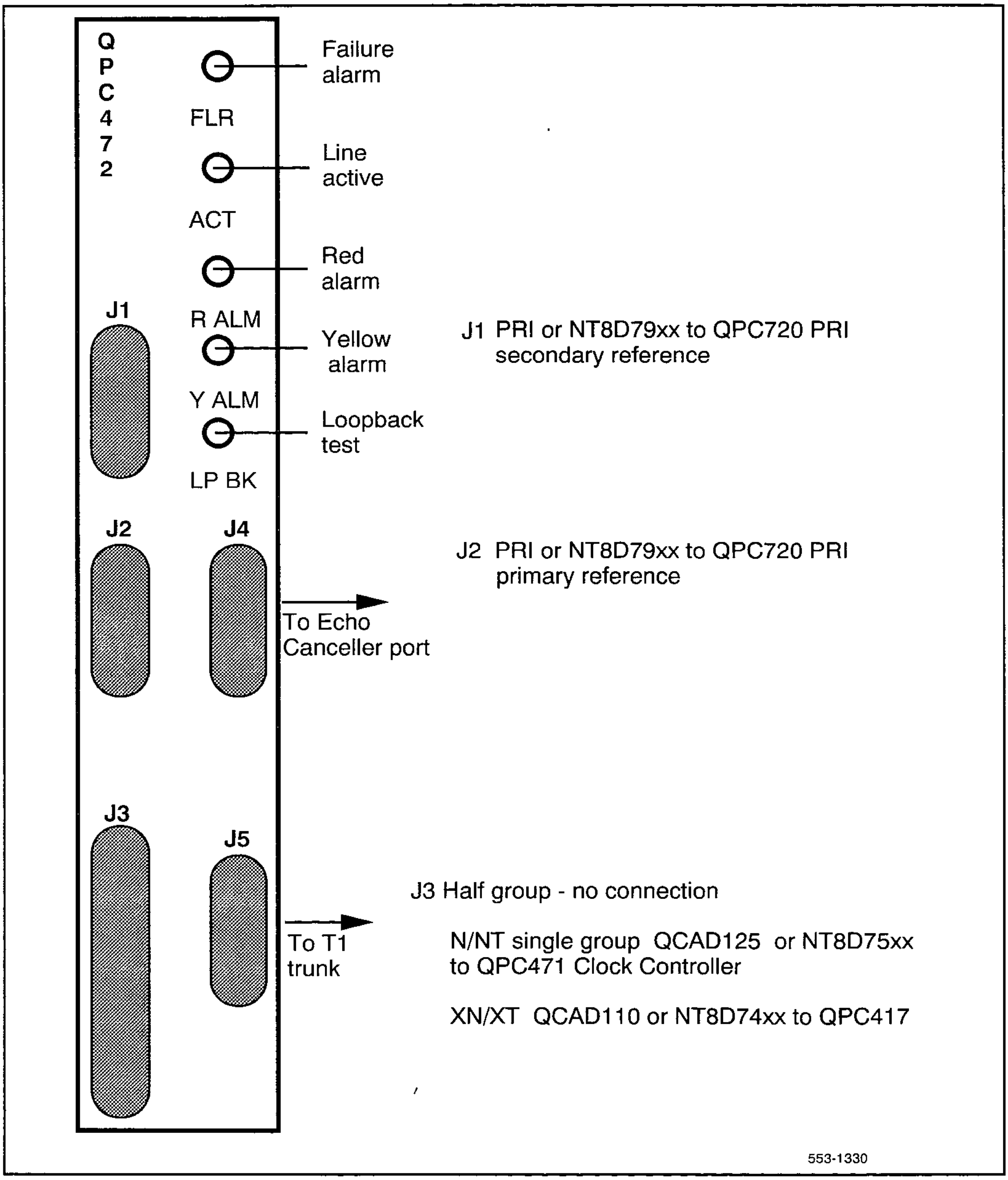
Table 1
QPC472 transmission equalization SW2 switch settings for DTI

Switch S2 settings	To repeatered facility	To cross-connect point
5 ON	0 - 45 m (0 - 150 ft.)	0 - 15 m (0 - 100 ft.)
2, 4, 6 ON	46 - 135 m (151 - 450 ft.)	16 - 100 m (101 - 355 ft.)
1, 3, 7 ON	136 - 225 m (451 - 750 ft.)	101 - 200 m (356 - 655 ft.)

Table 2
QPC472 DTI external connectors

Faceplate designations	Type
J1	9-pin female, D-connector
J2	9-pin female, D-connector
J3	36-pin connector
J4	15-pin female, D-connector
J5	15-pin male, D-connector

Figure 1
QPC472 DTI faceplate layout



tions

rd can replace the QPC472 DTI. The two cards are
11 release 5 and later software.

quires power and ground from the backplane only. Power

peres

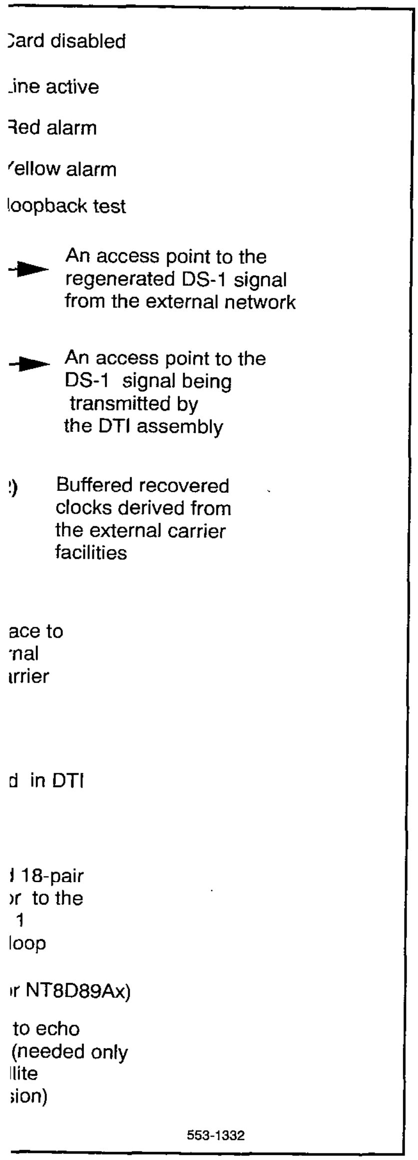
milliamperes

nilliamperes

ntains five LEDs and six external connectors.Table 3
ion equalization SW2 switch settings and Table 4 shows
5 lists the external connectors located on the QPC720
gure 2 shows the faceplate layout.

2 switch settings for PRI/DTI

eatered facility	To cross-connect point
m (0 - 150 ft.)	0 - 30 m (0 - 100 ft.)
m (151 - 450 ft.)	31 - 100 m (101 - 355 ft.)
5 m (451 - 750 ft.)	101 - 200 m (356 - 655 ft.)



Cable requirements

Shielded 22 AWG (0.644 mm) cable is recommended for connecting the DTI to the cross-connect point (or a CPI to CPI connection). This cable consists of two shielded twisted-pair conductors and two shields.

The transmit and receive pairs must be individually shielded and enclosed in a polyvinyl jacket. This type of cable is commonly referred to as "6-conductor" cable. T1 cable shielding should be grounded at the cross-connect point.

To install a private T1 transmission cable, consult the Northern Telecom Transmission Division. To manufacture cables of different lengths than the standard cables provided, see "Wire list" on page 57. Table 6 lists the types of cable used and the lengths required for internal and external DTI connections.

Table 6
DTI cables and cable lengths

Cable type	From	To	Maximum length (feet)	Maximum length (meters)
QCAD130 NT8D79xx	QPC472 QPC720	QPC471 QPC775 (CC-0)	10	3.04
QCAD130 NT8D79xx	QPC472 QPC720	QPC471 QPC775 (CC-1)	10	3.04
QCAD124 NT8D85xx	QPC472 QPC720	QPC414 Network	45	13.71
QCAD128 NT8D83xx	QPC472 QPC720	Bulkhead I/O panels	25	7.62
RS-232 NT8D89Ax	QPC472 QPC720	Echo Canceller	50	15.25
22AWG	Echo Canceller	DSX-1	655	199.64
NT8D97xx	I/O panel	T1, I/O	50	15.25

Installation procedures

The DTI/PRI card occupies two adjacent slots on a shelf. Locations vary as shown in the table below.

Table 7
Possible card locations by system

System	Shelf/Module	Slot
ST, STE	CE DTI expansion	5-12 2-8
MS	CE CE expansion	5-12 2-9, 11-12, 12-13
N/NT/RT	Network (LH) Network (RH) DTI expansion (RT only)	5-12 2-10 2-8
XN/XT	Network (LH) Network (RH)	4-10 5-11
Option 21A	CE/PE	Net: 4-8
Option 21	CE/PE	Net: 4-8 RPE: 1, 11-12
Options 51,61	CPU/Net	1-8 RPE:1, 11,12
Option 71	CPU	15-16 Net: 2, 5-9, 13-14 RPE: 1, 11-12

Table 8 shows the cable connections for the different systems. Not all cables apply to every system; read the Comments section carefully.

Table 8
Cable connections to the Clock Controller, Network, Patch panel, I/O panel, and Echo Canceller (Part 1 of 2)

Cable type	From	CON	To	DES	CON	Comments
QCAD130 NT8D79xx	QPC720/ QPC472	J1 J1	QPC471 QPC775	CC-0	J2	Run only when DTI is Primary reference clock source
QCAD130 NT8D79xx	QPC720/ QPC472	J1	QPC471	CC-0	J1	Run only when DTI is secondary reference clock source.
QCAD130 NT8D79xx	QPC720	J2	QPC471	CC-1	J2	Only when primary clock source. For RT in Single-Group mode only; not for ST, STE, or MS.
QCAD130 NT8D79xx	QPC720	J2	QPC471	CC-1	J1	Only when secondary clock source. For N/NT/RT in Single-Group mode only; not for ST, STE, or MS.
QCAD110 NT8D74xx	QPC471 QPC775	J3	QPC417	SCG0 SCG1		Applies only to XN/XT
QCAD125 NT8D75xx	QPC471	J3	QPC471	CC-1	J3	Applies only to N/NT/RT Single-Group mode.
QCAD124 NT8D85xx	QPC720/ QPC472	J3	Network			Run to connector on associated Network card. (For Half-Group mode only for the MS; for Single-Group mode only for the N/NT/RT).
QCAD124 NT8D85xx	QPC720/ QPC472	J3	Network			Run to connector on associated Network card. Applies only to N/NT/RT Half-Group mode.

Table 8
Cable connections to the Clock Controller, Network, Patch panel, I/O panel, and Echo Canceller (Part 2 of 2)

Cable type	From	CON	To	DES	CON	Comments
QCAD133 NT8D83xx	I/O Panel		Patch panel			Run via cabinet I/O panel to cross-connect terminal or Echo Canceller from shielded system.
QCAD133 NT8D83xx	QPC720/ QPC472	J4 J5	Patch panel			Run via cabinet I/O panel to cross-connect terminal or Echo Canceller from non-shielded system.
QCAD129 NT9J93xx	QPC720/ QPC472	J6 J4	I/O Panel			Run via cabinet I/O panel to cross-connect terminal or Echo Canceller from shielded system.
RS232 NT8D89xx	I/O Panel		Echo Canceller			Run via cabinet I/O panel to cross-connect terminal or Echo Canceller from shielded system.
RS232 NT8D89xx	QPC720/ QPC472	J6 J4	Echo Canceller			Run to cross-connect terminal or Echo Canceller from shielded system.

Follow the steps below to install the card.

1 Determine the cabinet and shelf location of the card to be installed.

2 Unpack and inspect cards.

3 Set option switches on the QPC472 as shown in Table 1. For the QPC720, refer to the switch and connector information in Table 3 and Table 4.

4 Install the card into the appropriate slot as shown in Table 7, "Possible card locations by system," on page 11.

5 Install Network card.

6 If required, install I/O adapters in I/O panel.

7 Attach cables. Refer to "Cable requirements" on page 10 for general cable information. Table 6 lists the cables required and Table 8 describes cable connections. Illustrations of the cabling arrangement for specific systems appear in the figures in each section.

Note: The figures in each section show cabling arrangements for the QPC720 card. The equivalent connections on the DTI card are:

QPC720 J4 = QPC472 J5
QPC720 J6 = QPC472 J4

8 If required, install connecting blocks at MDF or wall mounted cross-connect terminal.

9 If required, designate connecting blocks at MDF or wall mounted cross-connect terminal.

10 If required, install Network Channel Terminating Equipment (NCTE).

11 Cross-connect DTI circuits.

12 Run DTI acceptance/verification tests.

13 Add related office data into system memory. Refer to the work order and "Data administration" on page 43.

Removing the DTI card

1 Disable Network Loop using DTI Maintenance LD60. The command is DISL loop.

2 Remove data from memory.

3 Determine the cabinet and shelf location of the cards to be removed.

4 Remove cross connections at MDF to wall-mounted cross-connect terminal.

5 Disconnect DTI cables at Echo Canceller and at carrier interface (for example, Office Repeater, NCTE equipment).

6 Tag and disconnect cables from card. Rearrange Clock Controller card cables if required.

7 Remove DTI and Network cards.

Note: In a dual system, if the other circuit of the dual Network card is in use, ignore Step 7 for the Network card.

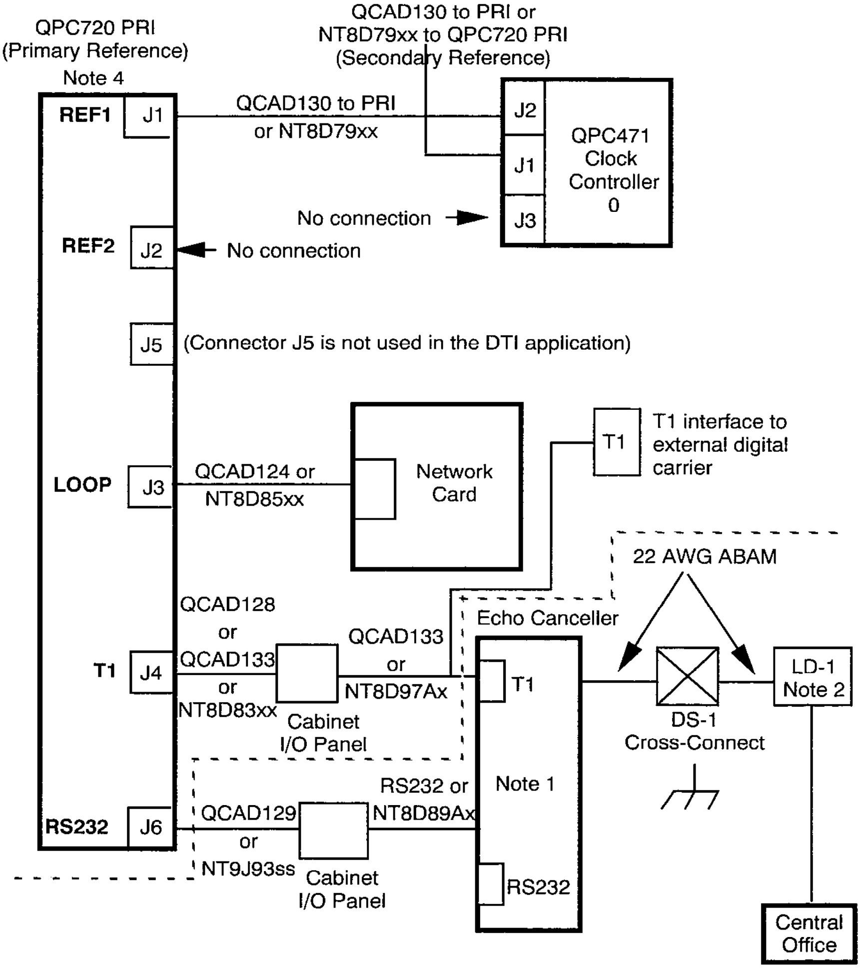
8 Pack and store card.

9 Close cabinet doors.

ST, STE systems

For basic information on installing a DTI card, refer to "Installation procedures" on page 11. See Figure 3 for DTI cabling arrangement in these systems. Figure 4 shows switch settings in DTI mode.

Figure 3
ST, STE cabling arrangement



Note 1: Maximum cable distance from DTI circuit packs to DS-1 cross connect is 655 ft.

Note 2: Maximum cable distance from DTI circuit packs to LD-1 is 750 ft.

Note 3: Connections below the dotted line are used only with an Echo Canceller.

Note 4: QPC720 does not interface with NT8D04 Superloop Network Card.

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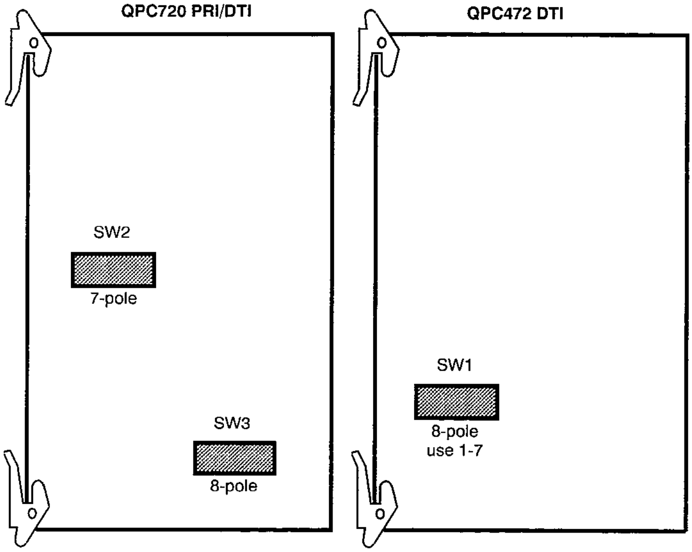
Figure 4
Switch settings in DTI mode



Note: See tables on PRI/DTI transmission equalization switch settings.

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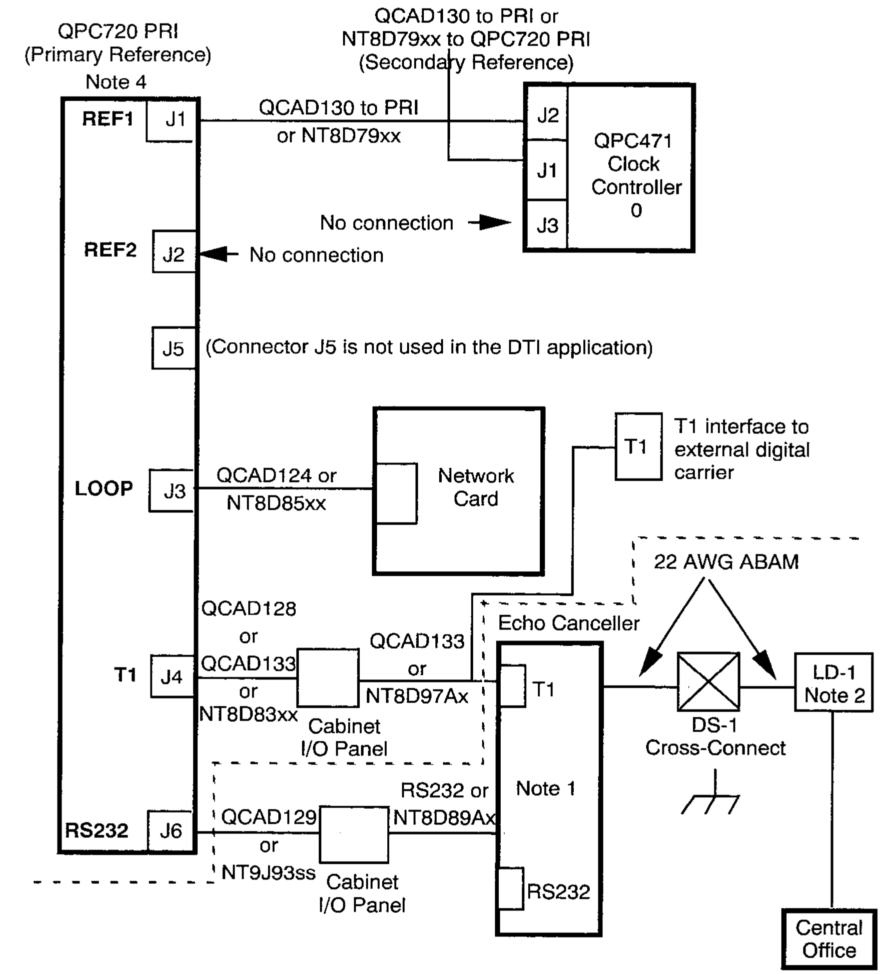
MS systems

For basic information on installing a DTI card, refer to "Installation procedures" on page 11. See Figure 5 for DTI cabling arrangement in MS systems. Figure 6 illustrates switch settings.

When no vacant positions are available to install DTI cards, additional Network shelves can replace Peripheral Equipment shelves located on the rear of the Common Equipment, Tape, or Disk shelves.

If an additional cabinet is required to install more Network shelves, use a QCA109 cabinet.

Figure 5
MS cabling arrangement



Note 1: Maximum cable distance from DTI circuit packs to DS-1 cross connect is 655 ft.

Note 2: Maximum cable distance from DTI circuit packs to LD-1 is 750 ft.

Note 3: Connections below the dotted line are used only with an Echo Canceller.

Note 4: QPC720 does not interface with NT8D04 Superloop Network Card.

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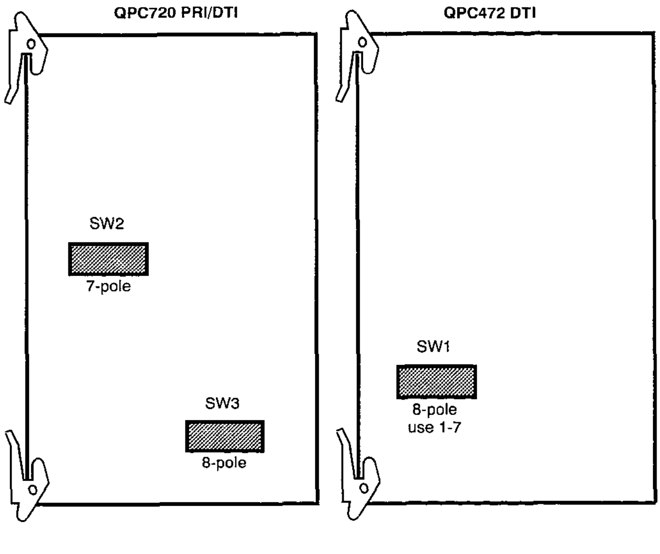
Figure 6
Switch settings in DTI mode



Note: See tables on PRI/DTI transmission equalization switch settings.

N/NT and RT systems

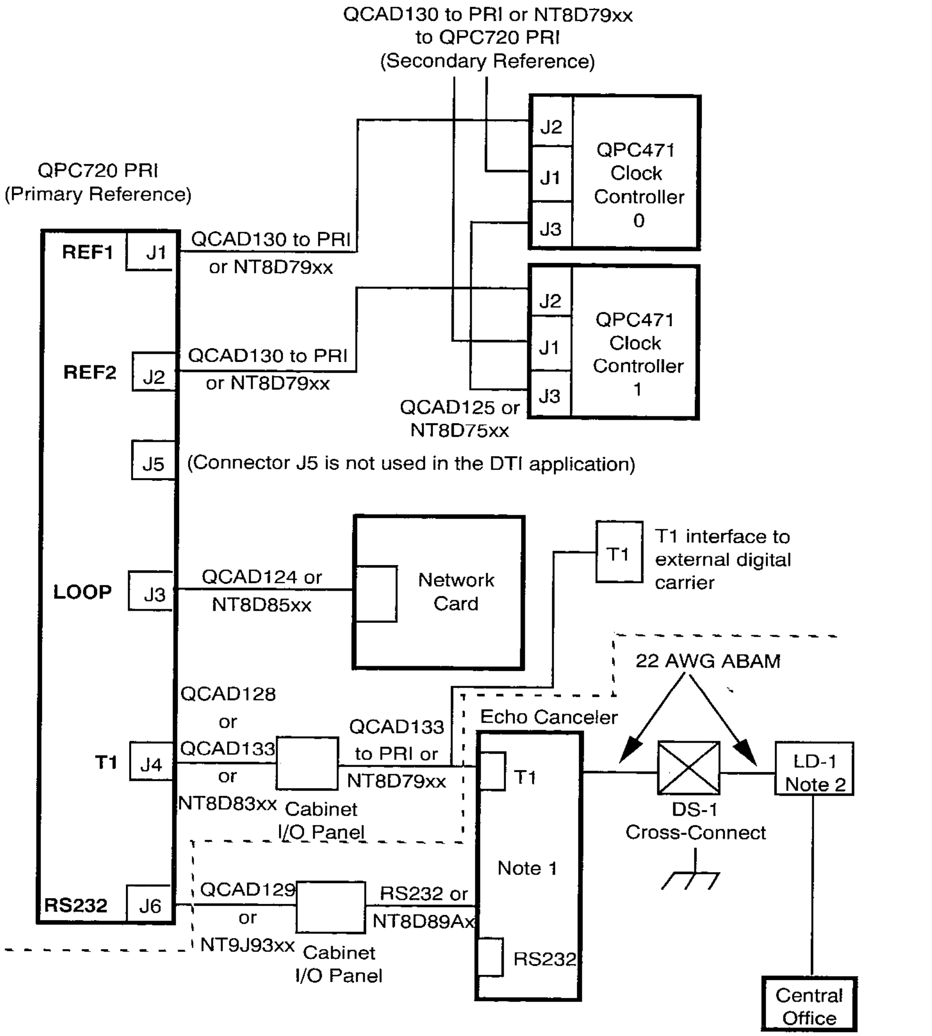
For basic information on installing a DTI card, refer to "Installation procedures" on page 11. As many as three cards can be plugged into an empty Network shelf, along with a Power Converter card.

When no vacant positions are available to install DTI cards, additional Network shelves can replace Peripheral Equipment shelves located on the rear of the Common Equipment, Tape, or Disk shelves. Refer to "Installing an additional Network shelf on the N/NT" on page 23 for details.

If an additional cabinet is required to install more Network shelves, use a QPC58 cabinet.

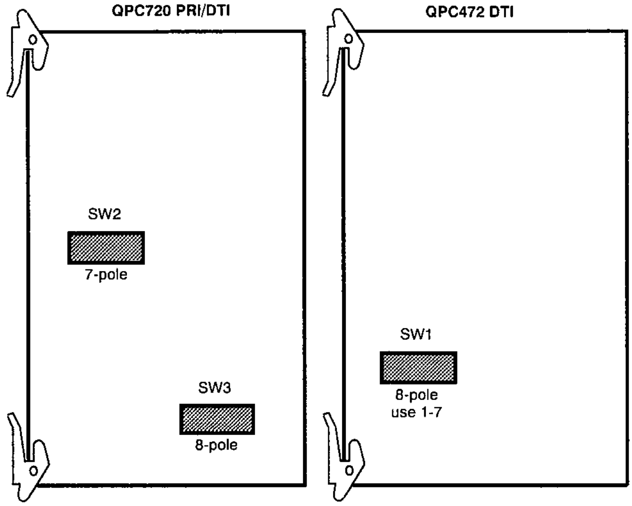
Figure 8 shows switch settings in DTI mode. See Figure 7 for DTI cabling arrangement in N/NT, RT, and 61 single and half group systems.

Figure 7
N/NT, 51, RT, and 61 single and half group cabling arrangement



Note 1: Maximum cable distance from DTI circuit packs to DS-1 cross connect is 655 ft.
Note 2: Maximum cable distance from DTI circuit packs to LD-1 is 750 ft.
Note 3: Connections below the dotted line are used only with an Echo Canceller.
Note 4: QPC720 does not interface with NT8D04 Superloop Network Card.

Figure 8
Switch settings in DTI mode



Note: See tables on PRI/DTI transmission equalization switch settings.

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Installing an additional Network shelf on the N/NT

> **CAUTION**
> The maximum number of DTI cards must not exceed three per shelf because the Network Shelf cable harness share the power breaker associated with each CPU.

This procedure is used when additional shelf space is required for DTI cards in N or NT, 51, and 61 systems. A QUD15 cooling unit is required for each additional shelf installed. When an additional shelf space is required it may be necessary to install a QCA108 network expansion cabinet to provide the additional space.

1 Determine the cabinet and shelf location of the cards to be removed.

Note: There are only two locations possible. They should be listed behind the MSU and CPU.

2 Unpack and inspect the shelf.

3 Remove the existing left or right rear PE shelf (if required).

4 Install the additional Network Shelf in the PE location. See Step 3

5 Install a QUD15 cooling unit directly below the Network Shelf and secure with four mounting screws.

6 Install and connect the QCAD172A power cable to the added QUD15 cooling unit as follows:

7 If the added QUD15 is located below the *left* Network Shelf, unplug the C11 connector from the QCAD111 power harness that connects to the existing left side QUD15.

8 Plug the C11 connector into the single-ended connector of the QCAD172A power cable.

9 Plug one of the two connectors at the other end of the C11 connector that was removed.

10 Plug the remaining connector of the QCAD172A power cable into the added QUD15.

11 If the added QUD15 is located below the ***right*** Network Shelf, use the C21 connector from the QCAD111 power harness instead of the C11. Otherwise, the procedure is identical.

12 At the QCAD111 power wiring harness, untie and then connect the C17 connectorized power connection cable to the right rear Network Shelf (use the C19 connectorized cable is connecting to the left rear Network Shelf).

CAUTION
Do not place the cards in the shelf until Step 12 is completed.

13 Install DTI trunks and enter related shelf and DTI office data into system memory.

XN/XT systems

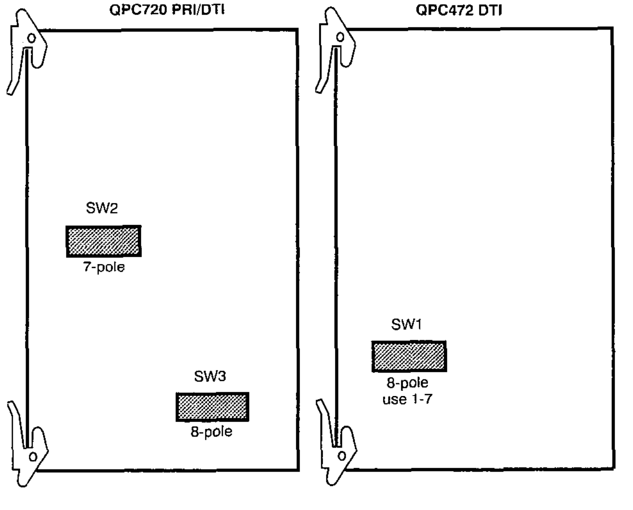
For basic information on installing a DTI card, refer to "Installation procedures" on page 11. As many as three cards can be plugged into an empty Network shelf along with a Power Converter card.

When no vacant positions are available to install DTI cards, additional Network shelves can replace Peripheral Equipment shelves located on the rear of the Common Equipment, Tape, or Disk shelves.

If an additional cabinet is required to install more Network shelves, use a QPC108 cabinet with XN or XT and 71 systems.

See Figure 10 for a diagram of the XN/XT/ 71 DTI cabling arrangement. Figure 9 illustrates switch settings.

Figure 9
Switch settings in DTI mode



Note: See tables on PRI/DTI transmission equalization switch settings.

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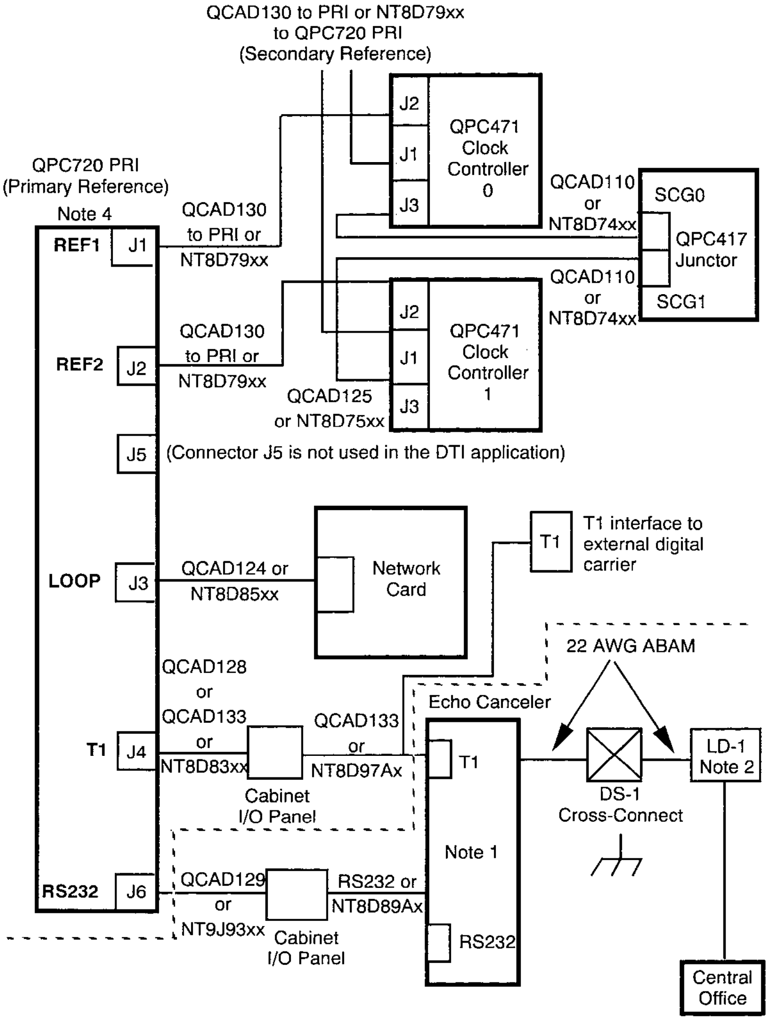
Figure 10
XN/XT and 71 cabling arrangement



Note 1: Maximum cable distance from DTI circuit packs to DS-1 cross connect is 655 ft.
Note 2: Maximum cable distance from DTI circuit packs to LD-1 is 750 ft.
Note 3: Connections below the dotted line are used only with an Echo Canceller.
Note 4: QPC720 does not interface with NT8D04 Superloop Network Card.

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21A, 21, 51, 61, 71, and 81

For basic information on installing a DTI card, refer to "Installation procedures" on page 11. For DTI cabling arrangements, refer to Figure 11.

Echo Canceller interface

An Echo Canceller is used in conjunction with a Meridian 1 system for satellite transmission. Refer to "Cable requirements" on page 10 for general cabling information. The Echo Canceller is controlled by an RS-232 port on the DTI card. Tables 9, 10, and 11 give the Echo Canceller pin assignments, operating parameters, and initialization procedures.

Table 9
DTI-to-Echo Canceller pin assignments

Signal	EIA RS232-C circuit designation	DTI pin	Echo Canceller pin
Transmitted Data (TXD)	BA	5	2
Received Data (RXD)	BB	2	3
Request to Send (RTS)	CC	-	4
Clear to Send (CTS)	CB	-	5
Signal Ground (common return)	AB	10	7
Received Line Signal Detector (DCD)	CF	1	8
Data Terminal Ready (DTR)	CD	4	20

Table 10
Operating parameters for the Echo Canceller

Data Transfer Rate	4800 baud
System Unit Number	1
Display Timeout	active
Failures Before Alarm	3

Table 11
Settings for the 24 channels on the Echo Canceller

Bypass	OFF
Off-Hook	ON
Canceller Only	OFF
H Reset	OFF
H Hold	OFF

Clock Controller Installation

There are three steps to installing a Clock Controller:

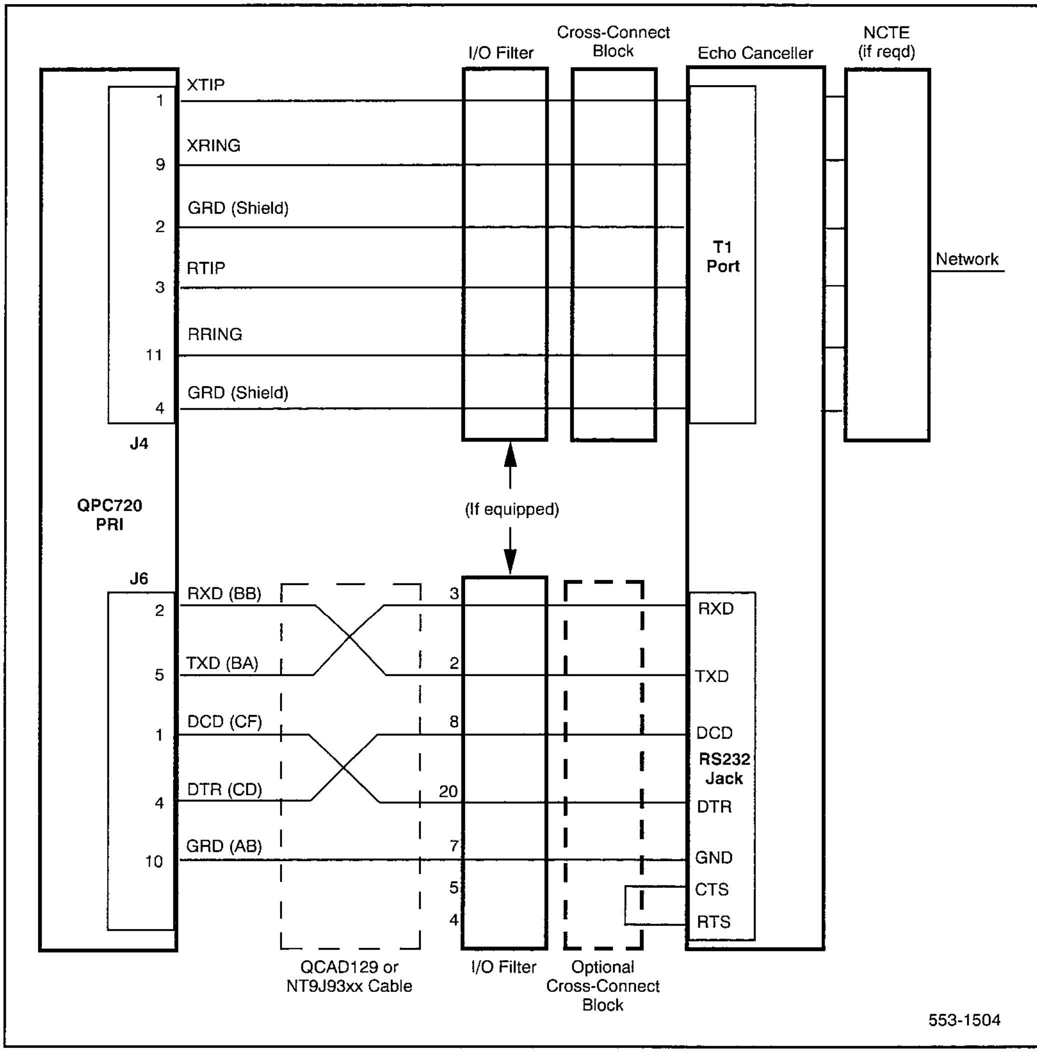
1 Determining the location of the Clock Controller card (shelf and slot).

2 Setting the switches on the card.

3 Inserting the card and connecting the cables. Refer to "Cable requirements" on page 10 for general cabling information. This step may also entail removing an old card.

The remaining sections of this chapter describe the details of this process.

Figure 11
DTI-to-Echo Canceller cabling



CAUTION
Do not deviate from the procedures described in this section, as deviation stops call processing.

Determining slots and shelves

The Clock Controller card installation site varies from system to system. Table 12 shows the systems, the shelves used, and the available slot or slots.

Table 12
Clock Controller shelves and slots

System	Shelf	Slot(s)
ST, STE	CE	5-12
MS	CE	9
RT	network	13
N, NT, 51	QSD39 network (LH) QSD40 network (RH)	13 2
XN	QSD17 CPU	14
XT	QSD62	15
21, 21E	NT8D11 CE	4-5
61	NT6D39 CPU/NET	9
71	NT8D34 CE Cube	14
81	core	6

Before installing a Clock Controller, set the switches as shown in Table 13, Table 14, Table 15, and Table 16 below. The first three tables display the settings for different vintages of the QPC471. Table 16 shows the settings for the QPC775.

Table 13
Clock Controller switch settings for QPC471 vintage A

System	Switch	Setting
N, NT, RT	SW2	ON
XN, XT	SW2	OFF
Vintage A applies only to these systems		

Table 14
Clock Controller switch settings for QPC471 vintages B, C, D, E, F, and G

System	Switch	Setting
ST, STE, 21, 21A, 21E	SW1 SW2 Jumper Plug J1 Jumper Plug J2	ON OFF TP8-9 TP11-TP12
MS	SW1 SW2 Jumper Plug J1 Jumper Plug J2	ON ON TP9-TP10 TP12-TP13
N, NT, RT, 51, 61	SW1 SW2 Jumper Plug J1 Jumper Plug J2	ON OFF TP8-TP9 TP11-TP12
XN, XT, 71	SW1 SW2 Jumper Plug J1 Jumper Plug J2	OFF OFF TP8-TP9 TP11-TP12

Table 15
Clock Controller switch settings for QPC471 vintage H

System	SW1				SW2				SW4			
ST, STE, 21A, 21, 21E	on	on	on	on	off	off	off	off	off	off	off	off
MS, SN	on	on	on	on	on	on	on	on	off	off	off	off
RT, N, NT, 51, 61	on	on	on	on	off	off	off	off	off	on	*	*
XN, XT, 71, 81	off	off	off	off	off	off	off	off	off	on	*	*
Cable length between the J3 faceplate connectors:												
0-4.3 m (0-14 ft.)											off	off
4.6–6.1m (15–50 ft)											off	on
6.4–10.1m (21–33 ft)											on	off
10.4–15.2 m (34–50 ft.)											on	on

• If there is only one Clock Controller card in the system, set to OFF. If there are two Clock Controller cards, set to match the cable length between the J3 faceplate connectors. Determine the total cable length (no single cable can exceed 25 ft.) between the J3 connectors. Both cards must have the same setting.

Table 16
Clock Controller switch settings for QPC775

System	SW2	SW3	SW4
N, NT, RT, ST, STE, 21, 21A, 21E, 51, 61	ON	OFF	ON
MS, SN	ON	ON	OFF
XN, XT, 71, 81	OFF	OFF	ON

Replacing a Clock Controller

Step 2 in the following procedures ("Installing a Clock Controller") explains how to replace a card.

Do not disable an active clock or a clock associated with an active CPU.

Installing a Clock Controller

Be sure to inspect the Clock Controller card before installing it. Refer to the tables earlier in this chapter for shelf, slot, and switch setting information. Remember not to use both the QPC471 and the QPC775 on a single system. QPC471 vintage H cards cannot be mixed with cards of an earlier vintage.

Starting the Clock Controller

The Clock Controller, when first enabled, is in free run mode. It stays in this mode for several minutes before being switched to tracking mode. Manual intervention via LD60 can force the Clock Controller to remain in a particular mode.

For the earlier QPC471 vintages, up to 20 minutes may pass before the clock actually locks and tracks. The QPC471 vintage G and H cards begin tracking within five minutes.

Clock Controller commands

During the installation procedure you will use some of the Clock Controller commands available in LD60. In the list of commands below, "x" refers to the Clock Controller number: 0 for the card associated with the CPU0, 1 for the card associated with CPU1.

- DIS CC x: Disable system Clock Controller x
- DSCK loop: Disable clock for specified loop
- DSYL loop: Disable yellow alarm processing for loop
- ENCK loop: Enable clock for specified loop
- ENL CC x: Enable system Clock Controller x
- ENYL loop: Enable yellow alarm processing for loop
- EREF: Enable automatic switchover of system clocks
- SSCK x: Obtain status of system clock x

- SWCK: Switch system clock between active and standby
- TRCK aaa: Set Clock Controller track where aaa can be PCK (the primary DTI/PRI reference), SCLK (the secondary DTI/PRI reference) or FRUN (free running)

Procedure 1
Installing the Clock Controller in the ST, STE, 21, 21E, MS, N, RT, 51, and NT half group

Note: Refer to the tables in this section to be sure you are using the correct vintage.

1 Set the ENL/DIS toggle switch to DIS (disable) on the new circuit card.

2 If replacing an existing card:

- Perform a status check on the clock with the SSCK command in LD60. The new controller should have the same status.
- Disable the old card using LD60.

Note: ERR20 messages may be generated. These can usually be ignored. However, excessive clock switching should be avoided, especially when counters are near the maintenance or out-of-service thresholds. Excessive switching could generate threshold-exceeded messages or cause the PRI to be automatically disabled. Check the counters in LD60. If necessary, reset the counters using the RCNT command.

- Set the old card's faceplate ENL/DIS switch to DIS.

3 Disconnect cables from the old Clock Controller card and remove it from the shelf.

> **CAUTION**
> The QCAD130 cable connecting the Clock Controller and DTI card must NOT be routed through the center of the cabinet past the power harness. Instead, it should be routed around the outside of the equipment shelves.

Note: The Clock Controller status display in this mode indicates NO UART (no universal async receiver transmitter). Do not perform a clock status check when receiving this code.

4 Install the new Clock Controller in the selected slot.

5 Connect the cables to the new card.

— Connect the Primary reference to J2.

— If applicable, connect the Secondary reference to J1.

6 Set the faceplate ENL/DIS switch to ENL (enable).

7 Set the error detection thresholds and clock synchronization control in LD73 (optional if replacing card; required with a new installation).

8 Enable the Clock Controller by entering ENL CCx in LD60.

9 To track on a primary or secondary reference clock, use LD60. The command:

TRCK	PCK	(for Primary)
	SCLK	(for Secondary)
	FRUN	(for Free-Run)

10 Issue the SSCK command to check controller status.

Installing the Clock Controller in a single group

Procedure 2
Installing the Clock Controller in the NT, 61

1 Set faceplate ENL/DIS switch to DIS (disable) on the new circuit card.

2 If replacing an existing card:

— Perform a status check on the clock with the SSCK command. The new card should have the same status.

— Disable the old card using LD60. Only a card in a standby mode associated with the standby CPU can be software-disabled.

Note: ERR20 messages may be generated. These can usually be ignored. However, excessive clock switching should be avoided, especially when counters are near the maintenance or out-of-service thresholds. Excessive switching could generate threshold-exceeded messages or cause the PRI to be automatically disabled. Check the counters in LD60. If necessary, reset the counters using the RCNT command.

— Set the old card's faceplate ENL/DIS switch to DIS.

— Disconnect the cables from the old Clock Controller card and remove it from the shelf.

Note: The Clock Controller status display in this mode indicates NO UART (no universal async receiver transmitter).Do not perform a clock status check when receiving this code.

3 If the 3PE switches have not been modified to recognize the Clock Controller card, adjust them as follows.

QSD39 (left hand side)SW1 OFF (Option 61 shelf 0)
SW2 ON
SW4 OFF

QSD40 (right hand side)SW1 OFF (Option 61 shelf 1)
SW2 ON
SW4 OFF
SW8 OFF

4 Set faceplate ENL/DIS switch to DISABLE.

5 Install Clock Controller in the selected slot.

6 Run and connect cables

- Connect the Primary reference to J2.
- If available, connect the Secondary reference to J1.
- Connect the cable between the two clocks to J3 on each controller card.

7 Set the faceplate ENL/DIS switch to ENL.

8 Enable the Clock Controller by entering ENL CC x in LD60.

9 Set the error detection thresholds and clock synchronization controls in LD73. (optional if replacing card; required with new installation)

10 To track on a primary or secondary reference clock, use LD60. The command:

TRCK	PCK	(for Primary)
	SCLK	(for Secondary)
	FRUN	(for Free-Run)

11 Issue the status check command, SSCK.

12 Optionally activate the new Clock Controller with the LD60 SWCK command.

13 Repeat, if necessary, for the second Clock Controller.

Procedure 3
Installing the Clock Controller in the XN, XT, 71, and 81

Note: The option 81 system requires a vintage H Clock Controller

1 Set faceplate ENL/DIS switch to DIS on new circuit card.

2 If replacing an existing card:

- Perform a status check on the clock with the SSCK command. The new card should have the same status.
- Disable it using LD60. (Only a card in standby state associated with the standby CPU may be disabled using software.)

Note: ERR20 messages may be generated. These can usually be ignored. However, excessive clock switching should be avoided, especially when counters are near the maintenance or out-of-service thresholds. Excessive switching could generate threshold-exceeded messages or cause the PRI to be automatically disabled. Check the counters in LD60. If necessary, reset the counters using the RCNT command.

- Set the old card's faceplate ENL/DIS switch to DIS.
- Disconnect the cables from the old Clock Controller card and remove it from the shelf.

Note: The Clock Controller status display in this mode indicates NO UART (no universal async receiver transmitter). Do not perform a clock status check when receiving this code.

3 Set the faceplate ENL/DIS switch to DIS.

4 Install the Clock Controller in the selected slot.

5 Run and connect the cables

- Connect the Primary reference to J2.
- If available, connect the Secondary reference to J1.
- Connect the cable between the two clocks to J3 on each controller card.

6 Set the faceplate ENL/DIS switch to ENL.

7 Enable the Clock Controller by entering ENL CC x in LD60.

8 Set the error detection thresholds and clock synchronization controls in LD73. (optional with card replacement; required with new installation)

9 To track on a primary or secondary reference clock, use LD60. The command follows.

TRCK	PCK	(for Primary)
	SCLK	(for Secondary)
	FRUN	(for Free-Run)

10 Issue the status check command, SSCK.

11 (Optional) Activate the newly installed Clock Controller with the LD60 SWCK command.

12 Repeat, if necessary, for the second Clock Controller.

Cabling requirements

Clock Controllers can require three different kinds of cable connections.

- In a single controller system, cable QCAD130/NT8D79xx connects QPC720/QPC472 (PRI or DTI card) to QPC471/QPC775 at J2 when DTI/PRI is the primary reference clock source.
- In a single controller system, cable QCAD130/NT8D79xx connects QPC720/QPC472 to QPC471/QPC775 at J1 when DTI/PRI is the secondary reference clock source.
- With dual controllers, cable QCAD125/NT8D75xx connects the Clock Controller cards to each other at J3 in single group mode.

NT8D74 Clock Controller to InterGroup cable

This cable connects the QPC471/QPC775 Clock Controller card to the NT8D36 InterGroup Module. It is available in the following lengths:

- NT8D74AC 1.2 m (4 ft.)
- NT8D74AD 1.8 m (6 ft.)
- NT8D74AE 2.4 m (8 ft.)
- NT8D74AF (QCAD110B) 3 m (10 ft.)
- NT8D74AJ 4.8 m (16 ft.)

NT8D75 Clock Controller to Clock Controller cable

This cable connects QPC471 Clock Controller cards to each other. It is available in the following lengths:

- NT8D75AC 1.2 m (4 ft.)
- NT8D75AD 1.8 m (6 ft.)
- QCAD125 3 m (10 ft.)

NT8D79 PRI/DTI to Clock Controller cable

This cable connects the PRI/DTI card to the QPC471 Clock Controller card. It is available in the following lengths:

- NT8D79AB 0.6 m (2 ft.)
- NT8D79AC 1.2 m (4 ft.)
- NT8D79AD 1.8 m (6 ft.)
- NT8D79AE 2.4 m (8 ft.)
- NT8D79AF (QCAD130) 3 m (10 ft.)

Data administration

This chapter describes the data administration procedures for the DTI feature. These include adding, changing, removing, and querying:

— the DTI feature

— the CPI application of DTI

The information presented here assumes the reader is familiar with the general principles of data administration on the Meridian 1 as described in *X11 software management* (553-3001-300).

DTI has an impact on the following Service Change and Print programs. For a complete description of these programs, see the *X11 input/output guide* (553-3001-400).

LD11	SL-1 Telephones
LD14	Trunks
LD16	Route
LD17	Configuration
LD20	Print Routine 1
LD21	Print Routine 2
LD22	Print Routine 3
LD25	Move Data Blocks
LD73	Digital Trunk Interface
LD81	Features/Stations Print

DTI data administration

A more detailed explanation of maintenance and out-of-service thresholds is located in *Digital Trunk Interface/Computer-to-PBX Interface maintenance* (553-2811-500). Also, refer to the *X11 input/output guide* (553-3001-400) for a complete listing and description of software programs.

Prompt	Response	Comments
Step 1	Gather data for the DTI thresholds and counters. Response to the following prompts in LD73 is required. Defaults are in parentheses.	
REQ	NEW, CHG, PRT, OUT, END	Action Request (create, modify, print, remove data, or terminate program)
TYPE	DDB	Digital data block
CEQT	Yes, No	Clock Controller equipped. Prompted for MS, ST, RT, N, NT, XN, or XT. Defaults to NO if response to REQ = NEW. Otherwise, defaults to previous value.
PREF	0-159	Primary reference source for clock synchronization controller. Use <cr> for free-running mode.
SREF	0-159	Secondary reference source for clock synchronization controller. Prompted only if PREF is not free-run. Use <cr> for free-running mode.
TRSH	0-15	Create, change, or remove a DTI threshold telephone. This number is entered in LD17 when defining the DTI loop. Use Xxx to remove TRSH (LD17 DLOP/TRSH associated with this TRSH must be removed first.) See Step 2.
RALM	1-(3)-128	Yellow (remote) alarm/clear threshold. This is the number of "remote alarm clear" signals received in 24 hours. If the threshold is reached, the DTI/PRI must be restored to service manually.
BIPC	0-(2)-128	24-hour bit rate violation threshold. If 0 is entered, there is no limit on number of times trunk can be taken out and automatically restored to service. See the *X11 input/ output guide* (553-3001-400).

Note: Automatic restoration is only available with Release 12 and later software.

Prompt	Response	Comments
LFAC	0-(3)-128	24-hour loss of frame alignment threshold. If 0 is entered, trunks will be restored to service automatically.
BIPV		Bit error rate (bipolar violation and CRC) monitoring limits
	1-(3)-4,	Maintenance threshold
	1-(2)-4	Out-of-service threshold
SRTK		Frame slip/tracking/monitoring limits
	1-(5)-24,	maintenance threshold, the minimum time, in hours, between slips
	1-(30)-3600	out-of-service threshold, the maximum number of slips per hour
SRNT		Frame slip/free run (non-tracking)/monitoring limits
	1-(15)-1024,	Maintenance threshold, the minimum time, in seconds, between 10 consecutive slips. When the threshold is reached, warnings are output to the maintenance TTY
	1-(3)-1024	Out-of-service threshold, the minimum time, in seconds, between 10 consecutive slips. When the threshold is reached, the DTI loop is taken out of service and a minor alarm is raised

Note: Automatic restoration is only available with Release 12 and later software.See *Digital Trunk Interface/Computer-to-PBX Interface maintenance* (553-2811-500) for a description.

Prompt	Response	Comments
SRIM	1-127	Improvement timer in minutes.
SRMM	1-(2)-127	Improvement criteria. If non-tracking mode maintenance threshold is exceeded SRMM or fewer timers in duration of SRIM, trunks are returned to service. If not, timer is restarted and monitoring continues.
LFAL	1-(17)-10240	Loss of frame alignment monitoring limits Maintenance threshold
	1-(511)-10240	Out-of-service threshold, the number of frame losses in 24 hours

Prompt	Response	Comments
Step 2	Gather data to add, modify, or remove a digital trunk loop. The following prompts in **LD17** require a response. Defaults are in parentheses.	
REQ	ADD, CHG, OUT	Action Request (add, change, or remove data)
TYPE	CFN	Configuration data block
CEQU	Yes, (No)	Change CE parameters
DLOP	loop dd ff	Digital Trunk loop options. lll = network loop number 0-159 dd = number of data calls 0-(24) ff = frame format D2, D3, D4, or ESF The default for frame format (ff) is ESF if prompt MODE is set to PRI; D3 if MODE is set to DTI or LINK.
Note: ESF can be specified for the QPC720 using X11 release 12 and later software.		
MODE	TRK, LINK	The mode in which the DTI hardware will be used. TRK = Digital Trunk mode, LINK = Digital Link mode
LCMT	B8S, AMI	The line coding method B8S = B8ZS binary 8 zero code substitution AMI = B7 zero substitution The default is B8S if the frame format is ESF; otherwise, the default is AMI.
YALM	DG2, FDL	The Yellow Alarm Pattern DG2 = Digit 2 FDL = Facility Data Link The default is FDL if the frame format is ESF; otherwise, the default is DG2.
TRSH	0-15	Digital Trunk threshold telephone. Enter the threshold telephone defined in LD73.
DTIC	0-159	The starting network loop slot occupied by the DTI card. It marks, as occupied, the loop slots taken up by the DTI card. On non-network loops, enter <cr> and ignore the message SCH 2035.
OVLY	Yes, (No)	Change overlay area options
SID	xxxx	System ID number The SID is used for polling a Meridian 1 for ACD, CDR, and Traffic reports.
DROL	60	Include DTI Hardware diagnostic in daily routines (LD60)

Prompt	Response	Comments
Step 3	Gather data to add, change, or remove a Digital Trunk route. Response to the following prompts in **LD16** is required. Defaults are in parentheses.	
REQ	NEW, CHG, OUT,	Action Request (create, modify, or remove data)
TYPE	RDB	Route date block
CUST	0-99	Customer number
ROUTE	0-511	Route number
	0-511[14+], 0-127[14+]	For System Options NT, RT, XN, ST, 51, 61, and 71 (R14) For all other machine types
TKTP	aaa	Trunk route type
	COT FEX WAT DID TIE	Central Office trunk route Foreign Exchange trunk route Wide Area Trunk route Direct Inward Dialing trunk route TIE trunk route
SAT	Yes, (No)	Trunk route via Earth orbiting satellite transmission. This prompt has no relation to the trunk route function connecting a main PBX to a satellite PBX.
DTRK	Yes, (No)	Digital trunk route; (must be Yes)
DSEL		Digital trunk route selection Prompted if response to TKTP is TIE and DTRK is YES
	DTA, VCE, TDN, (VOD)	Data only, Voice only, Transparent Data Networking, (Voice or Data route)

Prompt	Response	Comments
PTYP		Port type at far end
	Analog Tie trunk routes	
	AST	Analog satellite PBX TIE trunk or ESN satellite PBX TIE trunk
	(ATT)	Analog Tie trunks
	AOT	Analog Tie trunk, used instead of "ATT" whenever the PBX has one or more satellite trunk routes (dst) to any digital satellite PBX that includes OPX telephones.
	Digital Tie trunk routes	
	DCT	Combination satellite PBX TIE trunk
	DST	Digital satellite PBX Tie trunk
	(DTT)	Digital or combination Tie trunk
	Analog CO trunk routes	
	(ACO)	Analog CO trunk
	ATO	Analog Toll Office trunk
	Digital CO trunk routes	
	(DCO)	Digital or combination CO trunk
	DTO	Digital or combination Toll Office trunk
AUTO	Yes, (No)	Auto-terminate must be NO if response to DSEL was VOD
SIGO	aaa	ESN signaling arrangement. Prompted if the network signaling package is equipped, voice channels are configured VOD and response to TKTP was TIE. The allowed values are as follows:
	STD	Standard signaling arrangement
	ESN2	Supports NCOS, TCOS, and CCBQ call types
	ESN3	Supports Network Call Transfer, Satellite Link Control and all ESN2 call types.
	ESN5	Supports DTI data calls plus all other call types; Not allowed if response to AUTO is Yes
	EN19	Supports Transparent Data Networking and all ESN5 call types
	ETN	Provides Electronic Tie network signaling arrangement
STYP	SDAT, STDN	If Standard signaling arrangement (SIGO=STD) Uses Standard Data Signaling or Transparent Data Networking Signaling
MANO	Yes, (No)	Manual outgoing trunk route. Must be NO if response to DSEL was VOD or if response to SIGO was ESN5. Define the manual DN in LD14 (prompt MNDN).

Prompt	Response	Comments
Step 4	Gather data to add, change, move, or remove a Digital Trunk. Response is required to the following prompts in **LD14.** Defaults are in parentheses.	
REQ	NEW, CHG, MOV, OUT, END	Action Request (create, modify, move TN, remove data block, or terminate LD14)
TYPE	aaa	Trunk type
	COT FEX WAT DID TIE	Central Office Trunk data block Foreign Exchange trunk data block WATS trunk data block Direct Inward Dialing trunk data block TIE trunk data block See the *X11 input/output guide* (553-3001-400) for more options
TN	0-511, 1-24	Network loop, DTI channel
SFEX	Yes, (No)	Special digital FEX trunk (prompted only for digital FEX trunks)
TOTN	0-150, 1-24	Network loop, unassigned DTI channel; move to an unassigned channel on the same Network loop.
RTMB	0-127, 1-254	Route number (cannot be 31), member number
ATDN	xxxx	Auto-terminate DN at which the trunk is to automatically terminate. If response to DSEL in LD16 is VOD any valid DN is acceptable; If response is VCE, the DN must be a voice station (telephone or attendant); If response is DTA, the DN must be a data terminal.
MNDN	xxxx	Manual Directory Number. If response to DSEL in LD16 is VOD, any valid DN is acceptable; If response is VCE, the DN must be a voice station (telephone or attendant); If response is DTA, the DN must be a data terminal
SIGL	aaa	Trunk signaling for TIE or PAG trunks
	DX2, DX4, EAM, EM4, GRD, LDR, LOP, OAD	2-Wire and 4-Wire duplex E&M 2-Wire and E&M 4-Wire Ground Start, Loop Dial Repeating Loop Start, Outgoing Automatic incoming Dial
STRI	aaa	Incoming start arrangement
	DDL, IMM, OWK, WNK	Delayed dial, Immediate dial Off-hook Wink, Wink

Prompt	Response	Comments
STRO	aaa	Outgoing start arrangement
	DDL, IMM, OWK, WNK	Delayed dial, Immediate dial Off-hook Wink, Wink
SUPN	Yes, (No)	Answer and disconnect supervision required.
CLS		Class of service
	DTN, (DIP)	Digitone, (Dial Pulse)

Step 5 Gather data for each Meridian 1 telephone which is to be a data call telephone. Response is required to the following prompts in **LD11**. Defaults are in parentheses.

Prompt	Response	Comments
REQ	aaa	Action request
	CHG, CPY 1-32 END, MOV, MOV PAIR NEW, OUT	Modify existing data Copy a telephone template to add 1-32 new telephones Exit LD11, Move from one TN to another Move voice and data TNs Add new data to the system, Remove data block
TYPE	SL1	Define a Meridian 1 telephone or M11909 telephone
TN	lll s cc u	Terminal number: loop (0-159), shelf (0-3), card (1-10), unit (0-15) See prompt SFMT when using the copy command.
TOTN	lll s cc u	Destination TN. New loop, shelf, card, and unit when telephone data is to be moved to a new TN.
CLS	DTA, (VCE)	Data terminal, (Voice terminal) The default is DTA if a digital telephone is defined and TN unit is 8-15. The default is VCE if a digital telephone is defined and TN unit is 0-7.

CPI data administration

CPI is frequently connected to a computer by a direct cable. To restore service automatically with a direct connection, set all maintenance thresholds to their least sensitive values:

RALM – 128
BIPC – 128
LFAC – 128
BIPV – 1
SRTK – 1
SRNT – 1
LFAL – 10240

Note 1: If CPI uses a carrier facility requiring an NCTE or Clock Controller, follow the DTI instructions for NCTEs and Clock Controllers.

Note 2: For detailed information on setting Error Thresholds, refer to LD73 in the *X11 input/output guide* (553-3001-400).

Prompt	Response	Comments
1	Gather data to add, change, or remove a CPI Digital Trunk Loop. Response to the following prompts in **LD17** is required. Defaults are in parentheses.	
REQ	NEW, OUT, CHG, END	Add data, remove data, change data, or end LD17.
TYPE	CFN	Configuration data block
CEQU	Yes	Change CE parameters
DLOP	loop dd ff	Digital Trunk loop options. lll = network loop number 0-159 dd = number of data calls 0-(24) ff = frame format D2, D3, D4, or ESF The default for frame format (ff) is ESF if prompt MODE is set to PRI; D3 if MODE is set to DTI or LINK.
DTIC	0-159	The starting network loop slot occupied by the DTI card. It marks, as occupied, the loop slots taken up by the DTI card. On non-network loops, enter <cr> and ignore the message SCH 2035.
OVLY	Yes	Change overlay area options
SID	xxxx	System ID number. The SID is used for polling a Meridian 1 for ACD, CDR, and Traffic reports.
DROL	60	Include DTI Hardware diagnostic in daily routines

Acceptance tests

This chapter describes the acceptance test procedure for the DTI feature. This test verifies the operation of the system hardware.

The DTI card must be disabled before performing the self-test or call processing will be disrupted.

Self test procedures

Follow the steps below to activate the initial test procedure.

1 Log into the overlay area.

2 Enter **LD60**.

3 Enter **STAT L** to receive a status of the DTI card for Loop L.

4 Enter **DISL L** to disable DTI card for Loop L.

5 Enter **SLFT L** to test DTI card for Loop L.

6 If the system response not appropriate, refer to *Digital Trunk Interface/ Computer-to-PBX Interface maintenance* (553-2811-500) to analyze the message.

7 Enter **ENLL L** to enable the DTI card for Loop L.

8 Enter **** to abort LD60.

Automatic loop test

The automatic loop test checks the same functions as the self-test. Unlike the self-test, it can be run automatically as part of the midnight routines.

Set the ATLP command to 1:

- If all 24 channels are idle at midnight, Meridian 1 software disables the card and performs a self-test on all channels.
- If any of the 24 channels are busy at midnight, software disables one idle channel, chosen at random, and checks it while the card is enabled.

With the ATLP command set to 0, only one channel is tested. The channel tested is randomly selected by software, and it cannot be specified.

Automatic loop test procedures

Activate automatic loop test procedures with the following steps.

1. Log into the overlay area.
2. Enter **LD 60.**
3. Find an idle channel on the DTI loop by using the STAT command. Enter **STAT L**.
4. If all 24 channels are indicated BUSY by the response of the STAT command, this test cannot be run. Otherwise, select an idle channel and disable it. Enter **DSCH L C**.
5. Enter **ATLP 0** *or* **ATLP 1**.
6. If the system response is not appropriate, refer to *Digital Trunk Interface/ Computer-to-PBX Interface maintenance* (553-2811-500) to analyze the message.
7. Enter **ENLL L C** to enable DTI channel C.
8. Enter **** to abort LD60.

Link diagnostic and remote loopback tests

The remote loopback test and the link diagnostic test are performed manually on a per-channel or a per-frame (24 channels) basis.

Link diagnostic test The link diagnostic test, also called the far end, loopback test, does not test the Meridian 1 DTI. It puts the DTI in loopback mode so a remote loopback test can be performed on equipment at the far end.

The DTI channel or frame being tested must be disabled before performing the link diagnostic test or call processing will be disrupted.

Remote loopback test The remote loopback test, also called the near end, loopback test, checks the integrity of the DTI link from the Meridian 1 to the far end. The far end must be in loopback mode before this test can be performed.

The DTI channel or frame being tested must be disabled before performing the remote loopback test or call processing will be disrupted.

Coordinating the tests When a technician at the far end asks for loopback mode on the Meridian 1, enter:

LD 60
DISL loop
RLBK loop

To run the remote loopback test on the Meridian 1:

1 Call a technician at the far end. Ask for loopback mode at that facility.

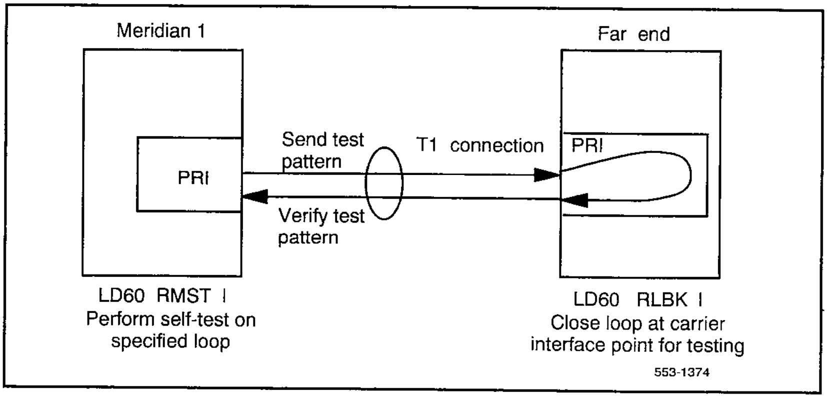
2 When loopback mode at the far end is confirmed, enter:

LD 60
DISL loop
RMST loop

Figure 12 shows the relationship between the remote loopback test and the link diagnostic test.

The procedure below is for performing a remote link diagnostic test on the DTI hardware when the DTI loop is disabled. This procedure assumes that the testing is being run from a local DTI to a collocated DTI.

Figure 12
DTI remote loopback test and DTI link diagnostic test



Link diagnostic and remote loopback test procedures

1 Log into the overlay area for both the local and collocated Meridian 1 systems.

2 Enter **LD 60**.

3 Enter **DISL loop** to disable the DTI card on the local Meridian 1 on Loop L. Enter **DISL loop** to disable the DTI card on the collocated Meridian 1 on Loop L.

4 On the collocated Meridian 1, enter **RLBK loop** to place this DTI in the remote loopback mode.

5 On the local Meridian 1, enter **RMST loop** to test the physical link between the local and the collocated DTIs.

6 If the system response is not appropriate, refer to *Digital Trunk Interface/ Computer-to-PBX Interface maintenance* (553-2811-500) to analyze the message.

7 On the collocated Meridian 1, enter **DLBK loop** to disable the remote loopback mode.

8 Enter **ENLL loop** to enable the DTI card on the local Meridian 1 on Loop L. Enter **ENLL loop** to enable the DTI card on the collocated Meridian 1 on Loop L.

9 Enter **** on both the local and collocated Meridian 1 systems to abort LD60.

Wire list

This chapter provides information required to build cables of nonstandard lengths for Digital Trunk Interface (DTI) applications. For installation of a private T1 transmission cable, consult Northern Telecom's Transmission Division.

The following pages contain lists and descriptions applying to Digital Trunk Interface (DTI) cables external to the Meridian 1 cabinet.

QCAD129/NT9J93xx

This cable is used to connect the RS-232-C interface for an Echo Canceller to the DTI card. Table 17 contains the QCAD129/NT9J93xx wire list.

Standard length	7 ft. (2.1 m)
Construction	22 AWG (0.644 mm), stranded
P1 Connector	15-pin, male, subminiature D, with jack-screws
P2 connector	25-pin, male, subminiature D, with jack-screws

Table 17
QCAD129NT9J93xx wire list

DTI signal	From	To	Echo Canceller signal
DCD	P1-1	P2-20	DTR
RXD	P1-2	P2-2	TXD
DTR	P1-4	P2-8	DCD
TXD	P1-5	P2-3	RXD
GND	P1-10	P2-7	GND
RTS	P1-12	P2-4	
CTS	P1-9	P2-5	
TPENBP1-15	nc		
Note: TPENB = Test Port Enable Bar.			

QCAD133/NT8D97xx

This cable is used to transport the T1 signal from the I/O filter assembly at the cabinet bulkhead to the Network Channel Terminating Equipment (NCTE) telephone company interface. For cabinets without an I/O filter assembly, this cable is used to transport the T1 signal from the QPC720 to the NCTE telephone company interface. Table 18 contains the wire list for the QCAD133/NT8D97xx.

Standard length	50 ft. (15.3 m)
Construction	Individually foil shielded, twisted pairs, 24 AWG (0.511 mm), stranded
P1 Connector	15-pin, female, subminiature D with jack-screws
P2 Connector	15-pin, male, subminiature D, with slide-latch. Optional spring-latch loose-packed with cable assembly.

Table 18
QCAD133/NT8D97xx wire list

Color	From	To	DTI signal
WHITE	P1-1	P2-1	XTIP to telephone company
BLACK	P1-9	P2-9	XRING to telephone company
GRN SHLD	P1-2	nc	GND
RED	P1-3	P2-3	RTIP from telephone company
BLACK	P1-11	P2-11	RRING from telephone company
RED SHLD	P1-4	nc	GND

QCAD128/NT8D83xx

This cable transports the T1 signal from the DTI card to the I/O assembly at the cabinet bulkhead. Table 19 shows the QCAD128/NT8D83xx wire list.

Standard length	10 ft. (3.05 m)
Construction	15-conducter ribbon, 28 AWG (0.321 mm), stranded
P1 Connector	15-pin, male, subminiature D, with jack-screws
P2 Connector	15-pin, male, subminiature D, with jack-screws

Table 19
QCAD128/NT8D83xx wire list

From	To	Signal
P1-1	P2-1	XTIP to telephone company
P1-2	P2-2	GND
P1-3	P2-3	RTIP from telephone company
P1-4	P2-4	GND
P1-5	P2-5	
P1-6	P2-6	
P1-7	P2-7	
P1-8	P2-8	
P1-9	P2-9	XRING to telephone company
P1-10	P2-10	
P1-11	P2-11	RRING from telephone company
P1-12	P2-12	
P1-13	P2-13	
P1-14	P2-14	
P1-15	P2-15	

SL-1

Digital Trunk Interface/Computer-to-PBX Interface

Installation and data administration

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SL-1

Digital Trunk Interface/ Computer to-PBX Interface

Maintenance

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This document is reissued to include technical content updates. Due to the extent of changes revision bars are omitted.

August 1, 1993

This document is reissued for updates and changes resulting from X11 release 19. All updates are noted with revision bars in the margins.

Contents

List of Figures

List of Tables

Introduction

Document overview

This document describes fault clearing and maintenance procedures for the Digital Trunk Interface (DTI) feature on Meridian 1. Fault clearing and maintenance procedures for the Computer-to-PBX Interface (CPI) application are the same as for DTI applications except for yellow alarm (remote alarm) recognition and transmission. (See "DTI fault clearing" on page 7).

This document covers the following topics:

— **Introduction** gives an overview of this document, lists related documentation, lists Meridian 1 programs impacted by the DTI feature, and provides a terminal number-to-channel number translation table.

— **DTI fault clearing** gives steps for performing a status check, clearing alarm conditions, and handling DTI problems.

— **DTI maintenance** covers maintenance commands, tests, system messages, error detection, the Error Counter, replacing a DTI pack, and cabling.

— **Clock Controller (CC) maintenance** describes Clock Controller operations, maintenance commands, system messages, replacing the Clock Controller, and cabling.

Information in this document is presented in a two-page modular format wherever possible. However, if the material requires only one page or more than two pages, the page between modules may be left blank.

Note 1: The DTI interface requires a QPC472 DTI or QPC720 Primary Rate Interface (PRI) card. Since this document deals with DTI functions, both cards are referred to as the DTI card or DTI hardware whenever their functionality is identical.

Note 2: Throughout this document, Meridian 1 refers to Meridian SL-1 M, N, NT, RT, ST, and XT machines and Meridian 1 system options 21, 51, 61, and 71, unless otherwise noted.

Other documentation

Refer to the following Northern Telecom Publications for more information:

– *X11 input/output guide* (553-3001-400)

– *Digital Trunk Interface/Computer-to-PBX Interface installation and data administration* (553-2811-200)

Affected programs

Maintenance Diagnostic Programs The DTI feature affects the following diagnostic programs, described in the *X11 input/output guide* (553-3001-400).

LD30 Network Signaling

LD32 Network and PE Replacement

LD36 Trunk Diagnostic No. 1

LD41 Trunk Diagnostic No. 2

LD45 Background Signaling and Switching

LD60 DTI Hardware

LD92 Automatic Trunk Maintenance

System Utility. DTI impacts two system utility programs, described in the *X11 input/output guide* (553-3001-400).

LD02 Traffic Control

LD80 Call Trace

Service Change Programs DTI impacts the following Service Change and Print programs, described in *Digital Trunk Interface/Computer-to-PBX Interface installation and data administration* (553-2811-200) and the *X11 input/output guide* (553-3001-400).

LD14	Trunks
LD16	Routes
LD17	Configuration
LD20	Print Routine 1
LD21	Print Routine 2
LD22	Print Routine 3
LD25	Move Data Blocks
LD73	Digital Trunk Interface
LD81	Features/Stations Print

TN to channel number conversion

DTI terminal numbers (TNs) have an equivalent channel number. The TN is output instead of the channel number in some Meridian 1 messages. The TN to channel number translation is shown below. Note that the translation is different for the D2 framing format than the formats for D3, D4 or Extended Superframe (ESF).

Terminal numbers (TNs) are identified in software by Loop (LL), Shelf (SH), Card (C), and Unit (U) numbers. Each channel on the DTI card is referenced by a TN. See Table 1.

Table 1
DTI channel numbers and equivalent terminal numbers (Part 1 of 2)

Channel Number	D2 Format TN SH C U	D3,D4,ESF Format TN SCHU
1	1 4 0	0 1 0
2	1 5 0	0 2 0
3	0 1 0	0 3 0
4	2 1 0	0 4 0
5	0 5 0	0 5 0
6	2 5 0	0 6 0
7	1 1 0	0 7 0
8	1 7 0	1 8 0
9	0 3 0	1 1 0
10	2 3 0	1 2 0
11	0 7 0	1 3 0
12	2 7 0	1 4 0
13	1 3 0	1 5 0
14	1 6 0	1 6 0
15	0 2 0	1 7 0
16	2 2 0	2 8 0
17	0 6 0	2 1 0
18	2 6 0	2 2 0
19	1 2 0	2 3 0

Table 1
DTI channel numbers and equivalent terminal numbers (Part 2 of 2)

Channel Number	D2 Format TN SH C U	D3,D4,ESF Format TN SCHU
20	2 8 0	2 4 0
21	0 4 0	2 5 0
22	2 4 0	2 6 0
23	1 8 0	2 7 0
24	3 8 0	3 8 0

DTI fault clearing

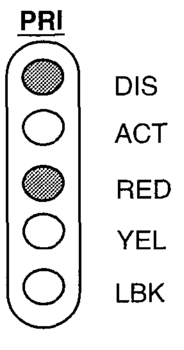
DTI status check

This status check is used to verify that the DTI card is working normally. It assumes the DTI hardware is properly installed (for example, correctly cabled) and operational. If the DTI status does not match the steps below, complete the check and proceed to DTI fault clearing procedures.

Note: The DTI interface requires a QPC472 DTI or QPC720 Primary Rate Interface (PRI) card. Since this document deals with DTI functions, both cards are referred to as the "DTI" card or "DTI" hardware whenever their functionality is identical.

1 Check the status LEDs on DTI cards.

For normal operation, only the ACT LED is lit (the LED on the QPC720 PRI is green; the LED on the QPC472 is red). Note if any other LED is lit and continue with the status check.



553-1340

2 Check the status of all DTIs:

LD60
STAT (L)

sample response:

DTI TRK LOOP L: ENBL
FFMT/LCMT/YALM: ESF/B8ZS/FDL
SERVICE RESTORE: YES
YELLOW ALARM PROCESS: YES
ALARM STATUS: NO ALARM
CH 1 IDLE DID
CH 2 BUSY TIE
.
.
CH 24

3 List DTI alarm counters:

LD60
LCNT (L)

Check the out of service counters to determine the number of out of service occurrences since last execution of the midnight routines.

sample response:

BVP xxxx
SLIPD xxxx
SLIPR xxxx
CRC xxxx
LOSFA xxxx
OS _BVP xxxx
OS_LOSF xxxx
OS_YEL xxxx

Red alarm (local alarm)

A DTI red alarm (local alarm) can indicate:

- T1 transmission problems
- a disabled DTI
- a faulty DTI card

Under any of these alarm conditions, all 24 channels are taken out of service as follows:

1 Meridian 1 software checks every 15 minutes to see if a Clock Controller or reference clock error has occurred. If the DTI in red alarm (local alarm) was a primary clock source, the software switches the Clock Controller to the secondary reference.

2 The DTI red alarm (local alarm) faceplate LED is lit.

3 Calls on the DTI are force disconnected.

4 The DTI card and all 24 channels are disabled.

5 After a pause of 2.5 seconds, the DTI sends a yellow alarm (remote alarm) signal to the far end.

6 Software sends the appropriate DTA error messages to the TTY.

Channel Restoration When the alarm condition improves, the DTI is restored to service as follows:

1 Red alarm (local alarm) is cleared.

2 After 10.5 ± 0.3 seconds, the DTI stops sending a yellow alarm (remote alarm) pattern to the far end.

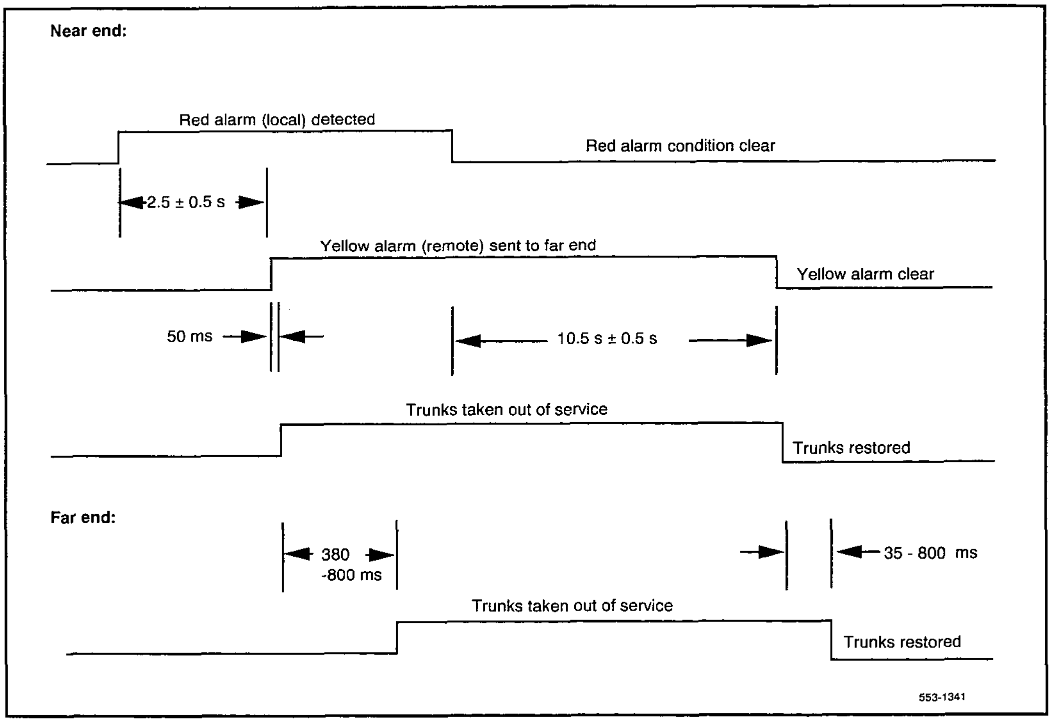
3 All channels but TIE trunks are placed into the idle state and made available for calls.

TIE trunks are made to match the state of the far end (as presented by the T1 port).

4 Software sends the appropriate DTA error messages to the TTY.

Figure 1 shows the time relationship between the red and yellow alarms at the local and far end DTIs.

Figure 1
DTI alarm timers



When a red alarm occurs

1. Check DTI status:

   LD60
   STAT (L)

2. Check DTI alarm counters:

   LD60
   LCNT (L)

3. See the chapter on DTI problems.

Yellow alarm (remote alarm)

A yellow alarm (remote alarm) on the Meridian 1 indicates a problem at the far end (the remote end). The fact that the DTI is receiving the yellow alarm pattern indicates that there is a T1 connection, but the far end is not ready.

> *Note:* For the Computer-to-PBX (CPI) application, yellow alarm (remote alarm) recognition and transmission are disabled in the DTI and are not provided at the host computer interface.

When the DTI receives the yellow alarm (remote alarm) signal from the far end, all 24 channels are disabled.

Channel Restoration When the DTI stops receiving the yellow alarm (remote alarm), the channels which were previously enabled and idle are returned to the idle state.

Each time a yellow alarm (remote alarm) is generated, a counter is incremented. When the yellow alarm 24-hour threshold (prompt RALM in LD73) is reached, the DTI can only be restored to service manually.

When a yellow alarm occurs

1. Do a status check.
2. Contact far end personnel to determine what action they are taking.

When the yellow alarm (remote alarm) 24-hour threshold is reached (DTA006 is printed):

1. Contact far end personnel to determine what action they are taking.
2. When the far end troubles are cleared, reset the alarm counters and disable, then enable the DTI card. To do this, use:

   LD60
   DISL L
   ENLL L

DTI problems

The DTI can have any of the following problems. Determine the cause of the problem and follow the recommended actions provided in Table 2.

Table 2
DTI problem solving

Symptom	Action
No connection to far end. If the T1 cable is not physically connected to the far end, frame alignment errors occur. The channels will be disabled, but the DTI will be in red alarm (local alarm) mode.	Use the Error Counter to verify the T1 transmission from the DTI faceplate (RCV and XMT) to each connection (cross-connect, LD-1 repeater, CSU, and such equipment).
DTI fails self-test.	Replace the DTI card.
Far end problems, usually indicated by a yellow alarm (remote alarm).	Do a DTI status check and contact personnel at the far end.
DTI T1 is connected but getting bit rate or frame errors. This can be caused by: — a bad T1 cable connection — electrical interference — carrier problems (for example, defective repeater or cross talk)	Use the Error Counter to verify the T1 transmission from the DTI faceplate (RCV and XMT) to each connection (cross-connect, LD-1 repeater, CSU, and such equipment).

Maintenance procedures

DTI commands (LD60)

DTI diagnostic commands are used to maintain both DTI and Clock Controller operation. The commands are organized as follows:

— card and channel commands (Table 3)

— test commands (Table 4)

— alarm and counter commands (Table 5)

Note: The DTI interface requires a QPC472 DTI or QPC720 Primary Rate Interface (PRI) card. Since this document deals with DTI functions, both cards are referred to as the "DTI" card or "DTI" hardware whenever their functionality is identical.

Self-test

The self-test, also called a local loopback test, checks speech path continuity, zero code suppression, remote alarm detection, and A&B bit signaling. This test is performed manually, on a per channel or a per frame (24 channels) basis.

The DTI card must be disabled before performing the self-test or call processing will be disrupted. To perform the self-test on a specific loop, use:

LD60
DISL L
SLFT L

Table 3
Card and channel commands (Part 1 of 2)

<table>
<tr><th>Command</th><th colspan="2">Description</th></tr>
<tr><td>DISI L</td><td colspan="2">DTI loop L is disabled only when all the channels are idle. The network and DTI card are then disabled and status LEDs are lit. Channel status is set to busy. Enter END to abort.</td></tr>
<tr><td>DISL L</td><td colspan="2">Disables network and DTI card of loop L. Active calls are force disconnected by on-hook simulation. All channels are disabled. Status LEDs DIS and RED (on QPC720) are lit.</td></tr>
<tr><td>DSCH L CH</td><td colspan="2">Channel CH of DTI loop L is disabled.</td></tr>
<tr><td>ENCH L CH</td><td colspan="2">Channel CH of DTI loop L is enabled. Channels other than TIE trunks are set to idle. TIE trunks are set to the same state as the far end (as presented by the T1 port).</td></tr>
<tr><td>ENLL L</td><td colspan="2">Enables DTI loop L. Channel CH of DTI loop L is enabled. Channels other than TIE trunks are set to idle. TIE trunks are set to the same state as the far end (as presented by the T1 port).</td></tr>
<tr><td>STAT</td><td colspan="2">Prints the status of all digital loops. The types are PRI (Primary Rate Interface), DTI (Digital Trunk Interface), and DLI (Digital Link Interface).</td></tr>
<tr><td rowspan="10">STAT L</td><td colspan="2">Prints status of DTI loop L. Refer to STAT L CH for channel status format. Loop status may be:</td></tr>
<tr><td>DTI TRK LOOP L</td><td>Network loop L is configured as a DTI trunk.</td></tr>
<tr><td>DSBL</td><td>DTI hardware is disabled.</td></tr>
<tr><td>ENBL</td><td>DTI hardware is enabled.</td></tr>
<tr><td>RLBK</td><td>DTI hardware is in remote loopback mode.</td></tr>
<tr><td>DISI PENDING</td><td>DISI command (disable when idle) is in progress.</td></tr>
<tr><td colspan="2">SERVICE RESTORE:</td></tr>
<tr><td>YES</td><td>Loop will restore service automatically if alarm condition is removed.</td></tr>
<tr><td>NO</td><td>Loop can only be manually enabled.</td></tr>
<tr><td>TRACKING</td><td>System clock is tracked to this loop.</td></tr>
</table>

Table 3
Card and channel commands (Part 2 of 2)

Command	Description	
	YELLOW ALARM PROCESS:	
	YES	Yellow alarm (remote alarm) processing is enabled.
	NO	Yellow alarm (remote alarm) processing is disabled.
	ALARM STATUS:	
	NO ALARM	No alarm active.
	RED	Red alarm (local alarm) active. May have T1 transmission problem. Action: 1 List alarm counters (LCNT L command). 2 Look for DTA messages. 3 Go to the fault clearing chapter.
	YELLOW	Yellow alarm (remote alarm) active. The far end is not ready or in local alarm mode.
	WAITING	DTI hardware is not responding. The card either did not respond to a status check or did not respond when a red alarm (local alarm) was cleared. Action: 1 Check DTI status again. 2 Disable then enable the DTI card (DISL L and ENLL L commands).
STAT L CH	Prints status of channel CH for DTI loop L. Status may be:	
	IDLE	Channel is enabled and is idle.
	BUSY	Channel has an active call, or channel is in a lockout state (the far end is disabled).
	MBSY	Channel is maintenance busy. The associated channel on the far end is disabled.
	DSBL	Channel is disabled.
	UNEQ	Channel is not equipped. DTI channels are 1-24.

Table 4
Test commands

Command	Description
ATLP 1/0	Automatic loop test enable (=1) or disable (= 0) 1 = Loop test enable; this will cause the far end to raise and clear a yellow alarm (remote alarm). 0 = Run the partial loop test; there is no interaction with the far end loop (default value).
RLBK L	Closes the loop at the carrier interface point of the DTI so the far end can perform an external loop-back test. DTI loop L must be disabled because the test disrupts call processing.
DLBK L	Disables the remote loop-back test previously set by RLBK L. The loop remains disabled.
DLBK L CH	Disables the remote loop-back test previously set by RLBK L CH. The channel remains disabled.
RLBK L CH	Same as RLBK L, but performed on a specified channel CH. This channel must be disabled prior to issuing the request.
RMST L	Performs self-test on loop L, providing the far end is in the remote loop-back mode.
SLFT L	Invokes DTI self-test on loop L. The loop must be disabled because the test disrupts call processing.
SLFT L CH	Invokes partial DTI hardware self-test using channel CH of loop L.

Table 5
Alarm and counter commands

Command	Description
CDSP	Clears the maintenance display on active CPU to 00 or blank.
CMIN C	Clears the minor alarm indicator for customer C.
CMIN ALL	Clears the minor alarm indicators for all customers.
LCNT	Prints content of all alarm counters of all DTI loops.
LCNT L	Prints content of all alarm counters of DTI loop L. The counters are: BPV — bit error rate counter indicates both bipolar violation and ESF CRC counts (the later only if configured in ESF mode) SLIPD — Frame slip deletion counter SLIPR — Frame slip repetition counter CRC — Cyclic Redundancy Check (CRC) in DTI always shows 0 if configured in ESF mode LOSFA — Loss of frame alignment counter OS_BPV — 24-hour bit error rate counter OS_LOSF — 24-hour loss of frame alignment counter OS_YEL — 24-hour yellow alarm (remote alarm) counter
LOVF C R	Lists threshold overflows for the customer and route specified. The overflows are set when the resident trunk monitor outputs a diagnostic message. Defined thresholds are HOLD, ILLR, REPT, SEIZ and SVFL (see LD16).
RSET L CH	Resets the thresholds for DTI loop L, trunk channel CH.
RCNT	Resets all alarm counters of all DTI loops.
RCNT L	Resets all alarm counters of DTI loop L.

Automatic loop test

The automatic loop test checks the same functions as the self-test. Unlike the self-test, it can be run automatically, as part of the midnight routines.

With the ATLP command set to 1:

1 If all 24 channels are idle at midnight, Meridian 1 software disables the card and performs a self-test on all channels.

2 If any of the 24 channels are busy at midnight, software disables one channel, chosen at random, and checks it while the pack is enabled.

With the ATLP command set to 0:

1 Only one channel is tested.

2 The channel tested is randomly selected by software; it cannot be specified.

3 The test is performed whether the channel is in a busy or idle state.

To perform the remote loop-back test, use:

LD60
ATLP 1 or 0

Link diagnostic and remote loop-back tests

The remote loop-back test and the link diagnostic test are performed manually on a per channel or a per frame (24 channels) basis.

Link diagnostic test The link diagnostic test, also called the far end loop-back test, does not test the Meridian 1 DTI. It puts the DTI in loop-back mode so a remote loop-back test can be performed on equipment at the far end.

The DTI channel or frame being tested must be disabled before performing the link diagnostic test or call processing will be disrupted.

Remote loop-back test The remote loop-back test, also called the near end loop-back test, checks the integrity of the DTI link from the Meridian 1 to the far end. The far end must be in loop-back mode before this test can be performed.

The DTI channel or frame being tested must be disabled before performing the remote loop-back test or call processing will be disrupted.

Coordinating the tests When a technician at the far end is asked to place his DTI in loop-back mode (the link diagnostic test), he issues the following commands in LD60:

LD60
DISL L
RLBK L

To run the remote loop-back test on the near end (local) Meridian 1:

1 Call a technician at the far end. Ask for loop-back mode at that facility.

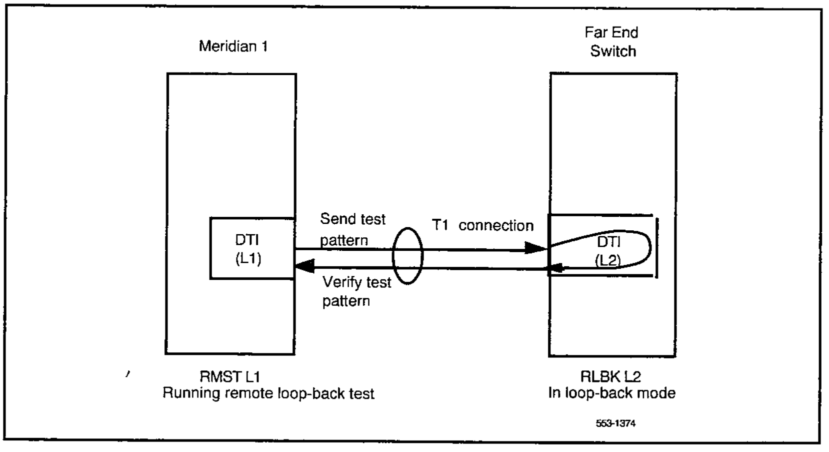
2 When loop-back mode at the far end is confirmed, the technician at the local end and issues the following commands in LD60:

LD60
DISL L
RMST L

After a brief time, there is an "OK" message.

Figure 2 shows the relationship between the remote loopback test and the link diagnostic test.

Figure 2
DTI remote loop-back test and DTI link diagnostic test



DTI system messages

There are two types of system messages – DTA messages (PRI/DTI alarm messages) and DTI messages (DTI/PRI error messages). See Table 6 for DTA messages; Table 7 lists DTI messages.

Table 6
PRI/DTI alarm messages (Part 1 of 2)

Message	Description
DTA001 L	Data block is not defined.
DTA002 L	Message received with wrong chip field.
DTA003 L	Power up message received.
DTA004 L	Phase lock loop (PLL) clear message is received without phase lock loop alarm.
DTA005 L	Yellow alarm (remote alarm) has occurred.
DTA006 L	Yellow alarm (remote alarm) 24-hour threshold has been exceeded. Manual intervention is required.
DTA007 L	Yellow alarm (remote alarm) is cleared.
DTA008 L	Yellow alarm (remote alarm) is disabled.
DTA009 L	Phase lock loop alarm has occurred.
DTA010 L	Phase lock loop alarm has cleared.
DTA011 L	Bit error rate warning threshold.
DTA012 L	Bit error rate out of service limit.
DTA013 L	Too many bit error rate out of service occurrences in 24 hours.
DTA014 L	Bit error rate alarm cleared.
DTA015 L	Frame slip—tracking—maintenance limit.
DTA016 L	Frame slip—tracking—out of service limit.
DTA017 L	Frame slip—free run (non-tracking)—maintenance limit.
DTA018 L	Frame slip—free run (non-tracking)—out of service limit.

Table 6
PRI/DTI alarm messages (Part 2 of 2)

Message	Description
DTA019 L	Frame alignment maintenance limit.
DTA020 L	Frame alignment out of service limit.
DTA021 L	Frame alignment alarm persisted for 3 seconds.
DTA022 L	Frame alignment alarm has cleared for at least 15 seconds.
DTA023 L	PRI loop is up.
DTA024 L	System initiated (automatic, LD45 or LD60) self-test on PRI loop L failed. All channels are disabled, loop is put into red alarm (local alarm).
DTA025 L	System initiated (automatic, LD45 or LD60) self-test on PRI loop L passed. Channels were previously disabled due to self-test fault or a loop-level self-test. Channels are enabled and red alarm (local alarm) is removed.
DTA026	Frame slip—free run (non-tracking)—out of service limit is reached. Trunks remain out of service. Reset improvement timer.
DTA028	Frame slip rate—free run (non-tracking)—improvement criterion is not met. Keep trunks out of service. Restart the improvement timer.
DTA029	Frame slip rate—free run (non-tracking)—improvement criterion is met. Bring trunks back into service.

Table 7
DTI/PRI error messages (Part 1 of 8)

Message	Description
DTI000	LD60 identifier.
DTI001	Invalid input character.
DTI002	Invalid command.
DTI003	Incorrect number of parameters.
DTI004	Incorrect customer number.
DTI005	Invalid parameter.
DTI006	Loop specified is not a DTI/PRI loop.
DTI007	DISI request already active.
DTI008	DISI command is completed.
DTI009 loop ch	DTI/PRI channel failed hardware self test. For DTI009 L M E, the output data is L = loop M = N for NI microprocessor (see Table 7.1) M = C for CI microprocessor (see Table 7.2) E = error code for debug purposes.

Table 7
DTI/PRI error messages (Part 2 of 8)

Message	Description
	Table 7.1—Error codes for NI microprocessor (M=N) 00 = NI selftest has finished. 01 = Undefined messout received 02 = Problem with group 2 error handling (invalid level) 03 = NI to CI FIFO full (128 messages lost) 04 = CI-1 Micro failed to initialize on power-up 05 = NI group 1 error handling - undefined condition found 06 = Bad MESSOUT number 6 encountered. 07 = NI Messout queue fails. 08 = NI Messin queue full. 09 = NI priority Messin queue is full. 10 = Bad MESSOUT number 10 encountered. 11 = TN = 0 read from regular queue. 12 = TN = 0 read from priority queue. 14 = Bad TN associated with MESSOUT number 4 15 = Bad TN associated with MESSOUT number 15 50 = External RAM in range 880h-8EFH failed (MESSIN queue) 51 = Internal RAM test failed. 52 = Pad RAM test failed. 53 = External RAM test failed. 54 = 8253 timer/counter test failed. 55 = Slip counter test failed. 56 = Loopback of TS16 frame 0 failed. 57 = Loopback of non fas TSO failed. 58 = Echo test to CI-1 micro failed. 255 = Loss of NI FIFO synchronization (Stop byte = 0 not found).

Table 7
DTI/PRI error messages (Part 3 of 8)

Message	Description
	Table 7.2—DTI009 error codes for CI microprocessor (M=C) 03 = A complete message was not received from NI micro 128 = Message received by CI-1 through FIFO requested an undefined task. 129 = Message number 26 contained more than one bit set to indicate which of the ABCD bits is to be used for PPM. 130 = An attempt was made to set the flag to invoke the pulse timer for TS 0. 131 = A request for a task defined under Messout 30 has been received with the TN of TS 0. 132 = Attempt was made to enable outpulsing TS0 or TS16. 133 = A Messout 31 has been received for TS 0 or TS 16 with the pulse hold time not = 0. 134 = An attempt has been made to set the bit to invoke the pulse timer for TS 0 or TS 16. 135 = A request for outpulsing was received, but outpulsing data was not downloaded. 137 = A request for PPM counting was received, but the was not downloaded. 138 = A Messout 30 was received requesting a task to be performed for DTI TS 16 which is not allowed. 139 = The 8031 on CHIP RAM failed selftest. 140 = The TS 16 signalling RAM and/or the TS 16 pick up buffer failed selftest. 141 = The CI-1 micro external RAM failed selftest. 142 = Attempt was made to set/clear the flag used to invoke PPM pulse timing for DTI timeslot 0 or 16. 143 = CI-2 micro responded to echo request message but response was in error. 144 = CI-2 micro failed to respond to request echo message. 145 = SL1 TN of MESSOUT received is not 0 0 7 3 (selftest) 146 = The TN of MESSOUT 26 received was not that or TS 0. 147 = The TN of MESSOUT 28 received was not that of TS 0. 148 = The TN of MESSOUT 29 received was not that of TS 0. 149 = Upon enabling the DTI pack, the CI-1 was unable to write Frame 0, TS 16 with '0B'. 150 = MESSOUT 26 was received with the PPM counting bit (abcd) all equal to zero.

Table 7
DTI/PRI error messages (Part 4 of 8)

Message	Description
	151 = MESSOUT 28 was received with the outpulsing bit (abdc) equal to zero. 152 = CI-2 failed to respond to the CI-1 watch dog message. 153 = The CI-2 failed to respond to five consecutive watchdog messages and is assumed to be out of service. 154 = Messout received requesting the lower nibble of MFAS pattern to be written with something other than '0000'. 155 = MESSOUT received requesting '0000' to be written into 'abcd' state. 156 = MESSOUT received with a TN outside the range shelf 0, card 8-1, unit 3-0. 157 = CI-1, NI FIFO overflowed, and has been cleared. 128 messages lost.
DTI010 loop c	System clock c cannot be switched in to replace active clock; or another loop is already in loopback mode.
DTI011 c	System clock c cannot be switched in because loss of service will result to the peripheral signaling pack listed.
DTI012 loop	Network card does not respond from IOTEST; loop is disabled.
DTI013 loop	No channel is available on loop for diagnostic test. No self-test is performed on this loop.
DTI014 loop	Loop is in remote loop-back mode; command not allowed.
DTI015 loop	Loop is not specified as primary or secondary clock reference source.
DTI016	The DTI package is not equipped; LD60 is not allowed to load.
DTI017	Previous command in progress; enter END to abort.
DTI018 loop	DISI command aborted.
DTI019	The loop/channel is not disabled for self-test or it is already in the requested set/reset loop-back mode.
DTI020 loop	Loop is already disabled. No action is taken.
DTI021	Attempt was made to disable input TTY loop.
DTI022 loop	Loop is already enabled or the Peripheral Signaling card is disabled.
DTI023 loop ch	Cannot disable/enable the specified channel.

Table 7
DTI/PRI error messages (Part 5 of 8)

Message	Description
DTI024	Loop is enabled but no response from hardware.
DTI025 loop ch	Terminal is not equipped.
DTI026 c r	Invalid input parameter to LOVF command for customer c, route r.
DTI027 loop	Loop is not in loopback test mode.
DTI028 loop ch	No test result received before timeout from the specified loop or channel.
DTI029 loop	Loop is enabled but red (local) and yellow (remote) alarms exist.
DTI030 loop	Loop is enabled but red alarm (local alarm) exists.
DTI031 loop	Loop is enabled but yellow alarm (remote alarm) exists.
DTI032 loop	Loop is in yellow alarm (remote alarm) state or is waiting for "yellow alarm sending ceased" message from hardware. Do not perform the automatic self-test.
DTI033 loop	Loop is in red/yellow/audit state. Command not allowed.
DTI034 loop	Switching of clock controller is not allowed for this machine type.
DTI034 loop m	Loop microprocessor m failed echo message selftest; m = N for NI microprocessor, m = C for CI microprocessor.
DTI035	Clock controller does not exist.
DTI035 loop ts	Network map in software indicates that timeslot ts of network loop is idle but the connection memory word for that slot on network pack is not idle. Probably a software fault
DTI036 loop	Continuity checker on loop is faulty. Network pack probably faulty.
DTI037 loop	Unable to read partial alarm counts from DTI hardware on loop. Following alarm counts are not complete.
DTI038 loop	No channel is available on loop for diagnostic. Self-test was not performed.
DTI038 loop	Loop not responding. Check enable switch on Network card (P1).
DTI039 loop	Continuity test failed on loop.

Table 7
DTI/PRI error messages (Part 6 of 8)

Message	Description
DTI040 loop ch	Loopback test failed on loop and channel.
DTI041 loop	Network memory test failed. Replace network pack.
DTI042 loop ch	No timeslots available for loopback test. Loopback test not tested on channel. If loop level test, all channels greater than and including channel ch not tested.
DTI043	Another channel already in loopback mode.
DTI044 loop	Loop enabled by midnight routine.
DTI045 loop	Self-test not performed on loop because loop was disabled manually.
DTI046 loop	Self-test not performed on loop because unable to access the loop.
DTI047 loop	Self-test not performed on loop since loop in remote loop back mode.
DTI050 loop	Continuity checker on loop is faulty. The network pack is probably faulty.
DTI051	Data link is not defined.
DTI052	Tracking rejected. Reference primary is not specified.
DTI053	Unable to track on primary.
DTI054	Tracking rejected. Reference secondary is not specified.
DTI055	Unable to track on secondary.
DTI056	Unable to access clock controller.
DTI057	Unable to free run.
DTI058	Supposed to free run but hardware is tracking on primary.
DTI059	Supposed to free run but hardware is tracking on secondary.
DTI060	Supposed to track on primary but hardware is tracking on secondary.
DTI061	Supposed to track on primary but hardware is free run.
DTI062	Supposed to track on secondary but hardware is tracking on primary.

Table 7
DTI/PRI error messages (Part 7 of 8)

Message	Description
DTI063	Supposed to track on secondary but hardware is free run.
DTI064	Cannot determine which CPU is in use.
DTI065	System clock must be switched before proceeding.
DTI066	Idle CPU must be switched in for active CPU before proceeding.
DTI067 c	System clock generator specified is already enabled.
DTI068 c	System clock generator specified is not responding.
DTI069 loop	Unable to track on Loop.
DTI070	Clock cannot be switched. Unable to track the reference loop.
DTI071	The ENCH L C command is not allowed here because the channel is busy.
DTI072	The ENCH L C command is not allowed because the associated D-channel is not established.
DTI073 loop	If loop is a QPC720, then the pack is not responding. If loop is a QPC472, then ignore this message.
DTI081	B-channel cannot be enabled until a DCH Link is established.
DTI098	Command entered applies to 1.5 Mb DTI only.
DTI099	Command entered applies to 2.0 Mb DTI only.
DTI100 loop	DTI link loop is associated with an indirect command and Status Link. This loop cannot be disabled until the CSL is disabled.
DTI101	Server using channel for maintenance, cannot Remove Link.
DTI202 L	DTI loop with line coding method of B8ZS cannot be enabled because it requires a QPC720 pack. Replace a QPC472 with a QPC720.

Table 7
DTI/PRI error messages (Part 8 of 8)

Message	Description
DTI203 L	DTI loop with Extended Super Frame (ESF) cannot be enabled because it requires a QPC720 pack. Replace a QPC472 with a QPC720.
DTI4130	Incompatible protocol between the interfaces. If this error continues, report it.
DTI4131	Incompatible protocol between the interfaces. If this error continues, report it.
DTI4132	ENCH L C command is not allowed because the associated D-channel is not established.

DTI error detection

Bit error rates

Bit error rate monitoring detects errors in transmission.

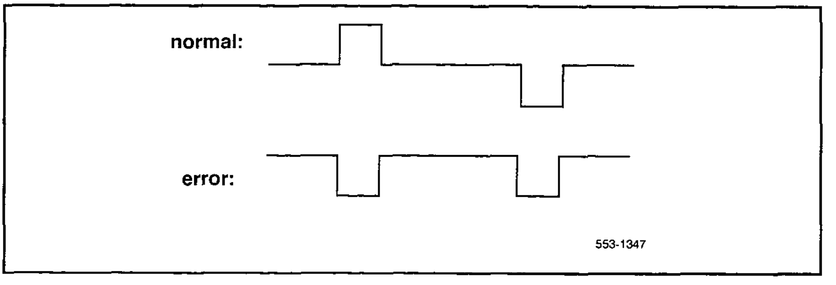
There are two methods of bit error rate monitoring: bipolar violation tracking and cyclic redundancy checks. With D2, D3, or D4 framing formats, only bipolar violation tracking is performed. With the Extended Superframe (ESF) framing format cyclic redundancy checks are performed.

Framing format is defined in

LD17
DLOP

Bipolar Violations (BPV) In a bipolar pulse stream, pulses alternate in polarity. If, after transmission, two pulses of the same polarity are received in succession (this could be caused by an electrical disturbance, such as noise), a bipolar violation has occurred. See Figure 3.

Figure 3
Bipolar violations



Cyclic Redundancy Check (CRC) The Extended Super Frame (ESF) format contains a checksum of all the data in the frame. The receiving side uses the checksum to verify the data is correct.

The primary difference between BPV and CRC is that bipolar violations indicate errors on the local span, while CRC indicates errors on an end-to-end span. For example, on a satellite link, BPV only detects errors in the span between the Meridian 1 and the satellite connection. Since CRC traverses the entire span, it indicates an end-to-end bit error rate.

In DTI mode with ESF framing format, the bit error rate checking is done as CRC. In DTI mode with D2, D3, or D4 framing formats, CRC error checking is not used.

Bit error rate thresholds DTI hardware detects BPV or CRC errors. It sends an overflow (OVFL) message to the Meridian 1 CPU each time 1024 BPV or CRC errors are detected. Running the midnight routines prints the number of overflows and clears the counter.

There are three bit error rate thresholds set in LD73. When a threshold is reached, a DTA message is output. See Figure 4.

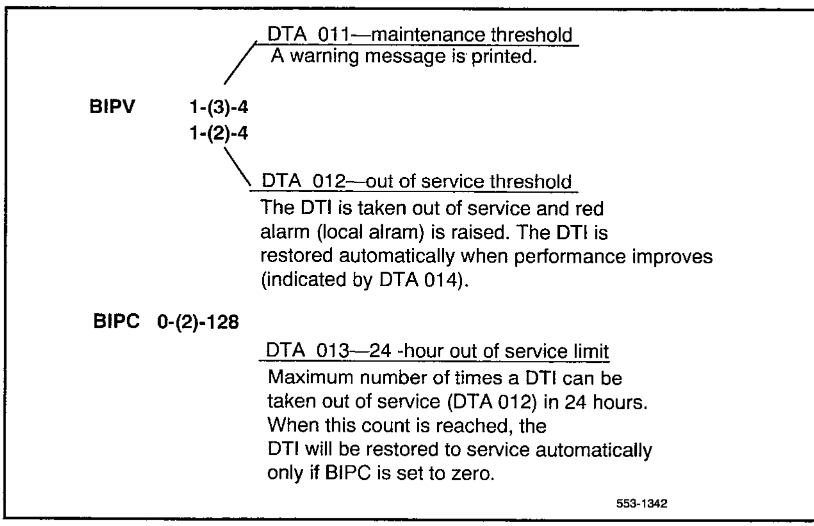
DTA011: Bit error rate maintenance threshold.

DTA012: Bit error rate out of service limit.

DTA013: Too many bit error rate out of service occurrences in 24 hours.

Note: Automatic restoration is available with Release 11 or later software.

Figure 4
BIPV and BIPC thresholds



The BIPV thresholds are based on the number of errors which occur in a given time. The BIPV threshold levels are shown in Table 8.

Table 8
BIPV thresholds

Level	Error rate	Elapsed time (seconds)
1	$>10^{-3}$	0.6639
2	$>10^{-4}$	6.639
3	$>10^{-5}$	66.39
4	$>10^{-6}$	663.9

For example, a level 3 error means that the error rat on the received transmission has exceeded one error in 100,000. Furthermore, this corresponds to no errors within a 66.39 second window of time.

Frame slip

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and demultiplexing operations. Monitoring frame slips detects frame deletion and repetition errors in clock synchronization.

Clock synchronization can be either tracking, on a primary or secondary reference clock, or free run (non-tracking). In LD73 (prompts PREF and SREF), one DTI may be defined as the primary clock reference. Another may be defined as the secondary clock reference. All others are defined as free run.

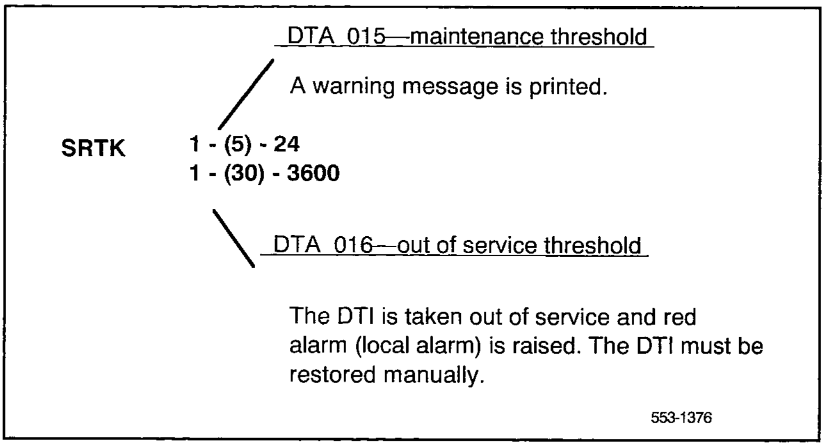
Tracking mode. DTI hardware detects frame slips in a tracking reference clock. Running the midnight routines prints the number of frame deletions and repetitions and clears the counters.

There are two thresholds set in LD73. See Figure 5. When a threshold is reached, a DTA message is output as shown below.

> DTA015: Maintenance limit for frame slips in tracking mode. The default is 1 slip in 5 hours.
>
> DTA016: Out of service limit for frame slips in tracking mode. The default is 30 slips in 1 hour.

Free run (non-tracking) mode. DTI hardware detects frame slips in the free run mode. Running the midnight routines prints the number of frame deletions and repetitions and clears the counters.

Figure 5
DTA messages



A maintenance threshold is set in LD73. An out of service threshold is also selected in LD73. When these thresholds are reached, DTA messages are output as shown below and illustrated in Figure 6:

DTA017: Maintenance limit for frame slips in free run (non-tracking) mode. The default is 10 slips in 15 seconds.

DTA018: Out of service limit for frame slips in free run (non-tracking) mode. The default is 10 slips in 3 seconds.

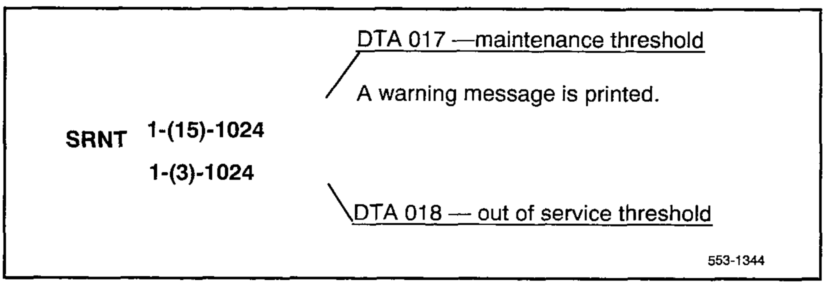
Note: With Release 12 and later software, an option for automatic recovery and an automatic recovery guard timer can also be set in LD73.

DTA026: Frame slip – free run (non-tracking) – out of service limit is reached. Trunks remain out of service. Reset improvement timer.

DTA028: Frame slip rate – free run (non-tracking) – improvement criterion is not met. Keep trunks out of service. Restart the improvement timer.

DTA029: Frame slip rate – free run (non-tracking) – improvement criterion is met. Bring trunks back into service.

Figure 6
DTA thresholds



Frame alignment

Loss of frame alignment monitoring detects out-of-frame conditions on the DS-1 bit stream.

Loss of frame alignment thresholds. DTI hardware detects out-of-frame conditions. Running the midnight routines prints the number of loss of frame alignment occurrences and clears the counters.

There are three frame alignment thresholds set in LD73. See Figure 7. When a maintenance or out of service threshold is reached, a DTA message is output as shown below.

DTA019: Frame alignment maintenance limit

DTA020: Frame alignment out of service limit

Note: Automatic restoration is available with Release 11 or later software.

Figure 7
Frame alignment thresholds

DTA 019 – maintenance threshold
A warning message is printed.

LFAL 1-(17)-10240
1-(511)-10240

DTA 020 – out of service threshold
The DTI is taken out of service and red alarm (local alarm) is raised. If the loss of frame alignment condition clears for at least 15 seconds, the DTI is restored to service.

LFAC 0-(3)-128

24 -hour out of service limit
Maximum number of times a DTI can be taken out of service (DTA 020) in 24 hours. When this count is reached, the DTI will be restored to service automatically only if LFAC is set to zero.

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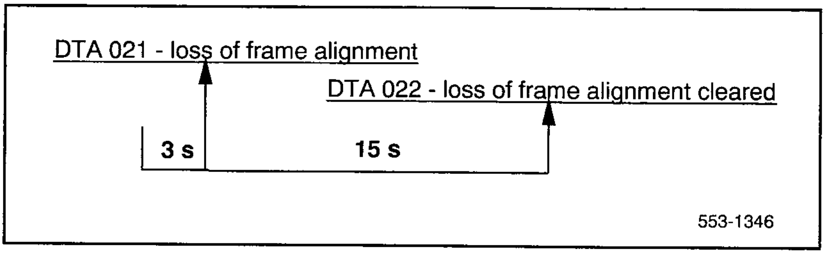
If a loss of frame alignment condition persists for three seconds, the affected DTI loop is taken out of service and a red alarm (local alarm) is raised. If the loss of frame alignment condition clears for at least 15 seconds, the DTI is automatically restored to service. The following DTA message is generated.

DTA021: Loss of frame alignment has persisted for 3 seconds. See Figure 8.

Figure 8
Loss of frame alignment



Using the Error Counter

The Error Counter detects bipolar violations or no-signal periods. It counts, stores, and displays them, to a maximum of 9999.

The DTI fault detection and isolation procedures described in this chapter are performed using an Option 81 portable test package, which consists of one each of the following items:

- Option 81 TTT2028 Mini-Error Counter, plus operation instruction card
- cord equipped with a bantam plug at one end and minihooks at the other
- loop-back plug (shorts pins 3 to 1 and 11 to 9 of a 15 pin D connector)

Follow these steps to use the Error Counter.

1 Plug one end of a patch cord into the input jack of the test telephone.

> **CAUTION**
> To prevent injury form voltage on the span, always connect the patch cord into the test telephone before connecting the other end to the external signal source.

2 Plug the other end of the patch cord into the monitor jack (RCV and XMT) of the QPC720 PRI or the apparatus associated with the QPC472 DTI being tested.

3 Monitor the Error Counter LED indicators described on the following page.

Error Counter switch functions

See Table 9 for Error Counter switch functions.

Table 9
Error Counter switch functions

Switch	Function
Display Enable	When held down, the switch enables the Counter display and the GOOD and O/R LED displays.
Reset	Zeros the counter.
Error/Error	Selects error counting seconds for bipolar violations or error-seconds.

Error Counter display functions

Table 10 lists display functions for the Error Counter.

Table 10
Error Counter display functions

Display	Function
GOOD	Indicates the presence of an acceptable bipolar signal (if bipolar violations, missing pulses or an oscillating line are detected, the indicator is off).
ERR	Flashes when bipolar violations are detected.
W/M	Indicates no input (absence of pulse) or an oscillating line.
O/R	Over range display turns on when the counter input has exceeded 9999 (the counter resets to 0000).
CNTR	With Error/Error-Second switch in the Error position, the unit counts errors at a maximum rate of 200 per second. With Error/Error-Second switch in the Error-Second position, the unit counts error seconds at a rate of one per second.

Replacing the DTI card

> **CAUTION**
> Firmly touch the metal frame of the cabinet to discharge static electricity from your body before handling cards.

1 Disable the DTI loop:

LD60
DISL L

2 Disconnect cables on DTI faceplate.

3 Remove the DTI from the shelf.

4 Ensure that the new DTI card switch settings are the same as the faulty DTI pack.

Figure 9 shows switch locations for the QPC720 and QPC472 cards. Tables 11 through 13 list transmission equalization switch settings for DTI.

5 Install the new DTI card in the appropriate slot.

6 Connect the network loop cable, the carrier interface cable, and the Echo Canceller cable. If the DTI pack is defined as a primary or secondary clock source, connect the Clock Controller cables(s) (see Figures 3 and 4).

7 Test the DTI card:

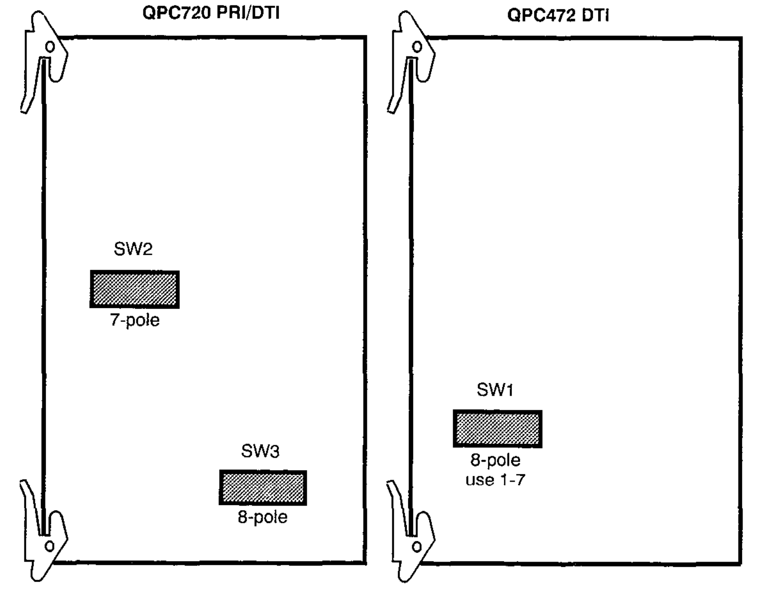
LD60
SLFT L

If an error message results, see the DTI fault clearing chapter.

8 Enable the DTI:

LD60
ENLL L

Figure 9
Transmission equalization switch settings



Note: See tables on PRI/DTI transmission equalization switch settings.

Table 11
QPC720 transmission equalization SW2 switch settings for PRI/DTI

Switch S2 settings	To repeatered facility	To cross-connect point
5 ON	0 - 45 m (0 - 150 ft.)	0 - 30 m (0 - 100 ft.)
2, 4, 6 ON	46 - 135 m (151 - 450 ft.)	31 - 100 m (101 - 355 ft.)
1, 3, 7 ON	136 - 225 m (451 - 750 ft.)	101 - 200 m (356 - 655 ft.)

Table 12
QPC472 transmission equalization SW2 switch settings for DTI

Switch S2 settings	To repeatered facility	To cross-connect point
5 ON	0 - 45 m (0 - 150 ft.)	0 - 15 m (0 - 100 ft.)
2, 4, 6 ON	46 - 135 m (151 - 450 ft.)	16 - 100 m (101 - 355 ft.)
1, 3, 7 ON	136 - 225 m (451 - 750 ft.)	101 - 200 m (356 - 655 ft.)

Table 13
SW3 options for the QPC720 PRI/DTI

Switch 3 options for PRI/DTI mode	
SW3-1	ON = extended superframe format (ESF) OFF = superframe format (SF)
SW3-2	ON = B8ZS line encoding (required for 64 kbs) OFF = AMI line encoding
SW3-3	ON = facility data link (FDL) yellow alarm method (cannot be used in Canada) OFF = Digit 2 yellow alarm method

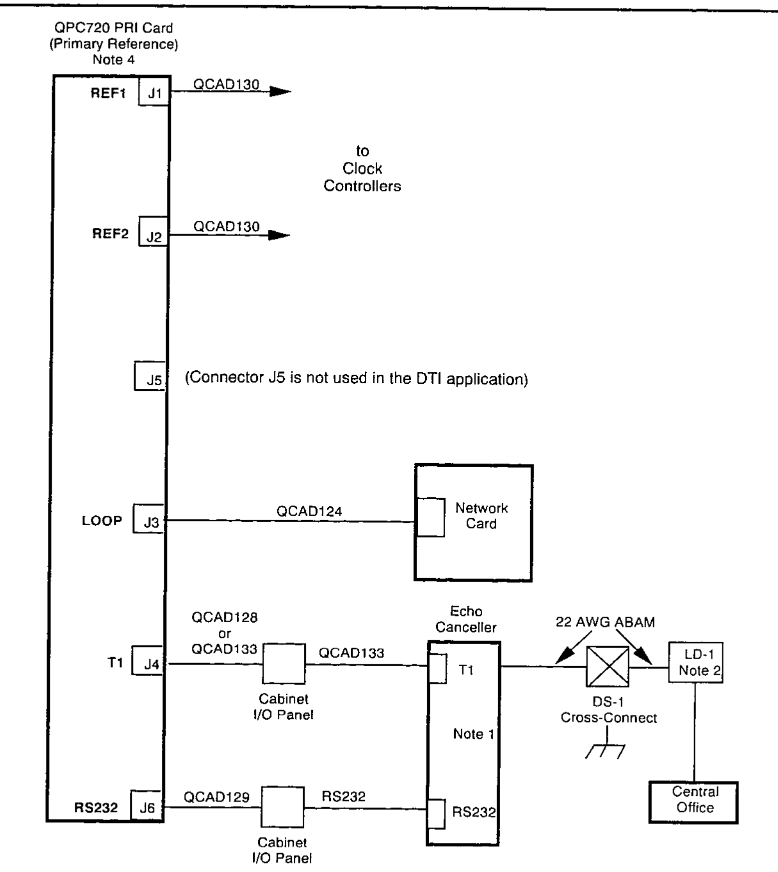
Note 1: All switch positions for S2 (location B22) are OFF except as shown under the column labeled "Switch S2 settings." The switch positions of the 8-pole SW3 (location E37) are OFF for DTI mode. For PRI mode, set SW3 positions as configured in LD17.

Note 2: For D2, D3, or D4 framing formats (superframe formats), set all SW3 options to OFF.

DTI cabling

Figures 10 and 11 illustrate DTI cabling with and without an Echo Canceller.

Figure 10
DTI cabling with Echo Canceller



Note 1: Maximum cable distance from DTI card to DS-1 cross connect is 655 ft.

Note 2: Maximum cable distance from DTI card to LD-1 is 750 ft.

Note 3: QPC720 does not interface with NT8D04 Superloop Network Card.

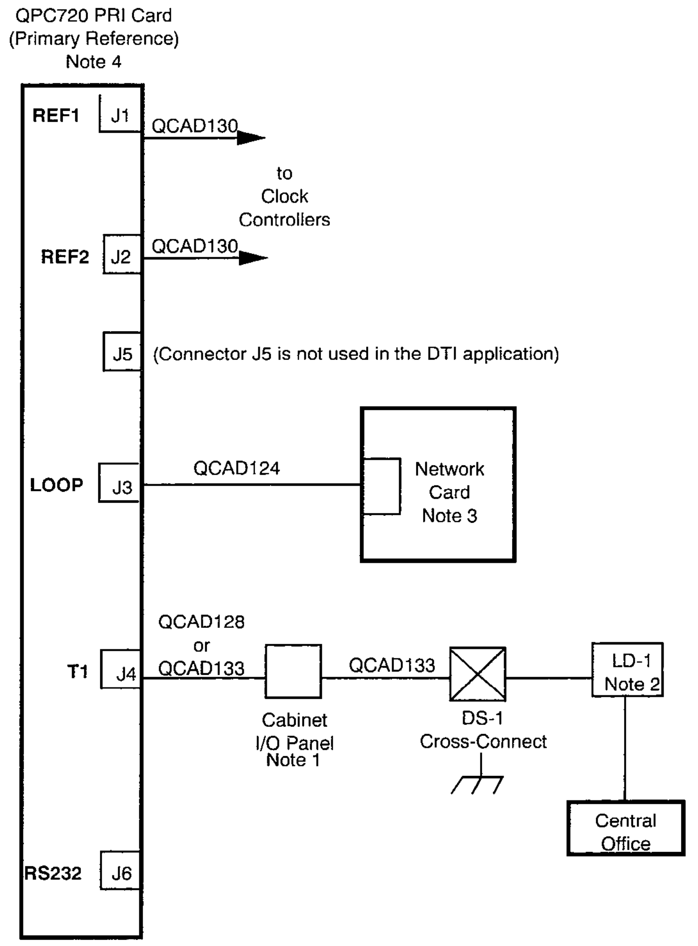
Note 4: Equivalent connections on the QPC 472 DTI card are:

QPC720 J6 = QPC472 J4

QPC720 J4 = QPC472 J5

553-1508

Figure 11
DTI cabling without Echo Canceller



Note 1: Maximum cable distance from PRI card to DS-1 cross connect is 655 ft.

Note 2: Maximum cable distance from PRI card to LD-1 is 750 ft.

Note 3: QPC720 does not interface with NT8D04 Superloop Network Card.

Note 4: Equivalent connections on the QPC472 DTI card are:

QPC720 J6 = QPC472 J4

QPC720 J4 = QPC472 J5

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Clock Controller maintenance

Clock operation

Tracking mode

In tracking mode, the DTI loop supplies an external clock reference to a Clock Controller. Two DTI loops can operate in tracking mode, with one defined as the primary reference source for clock synchronization, the othe defined as the secondary reference source. The secondary reference acts as back-up to the primary reference.

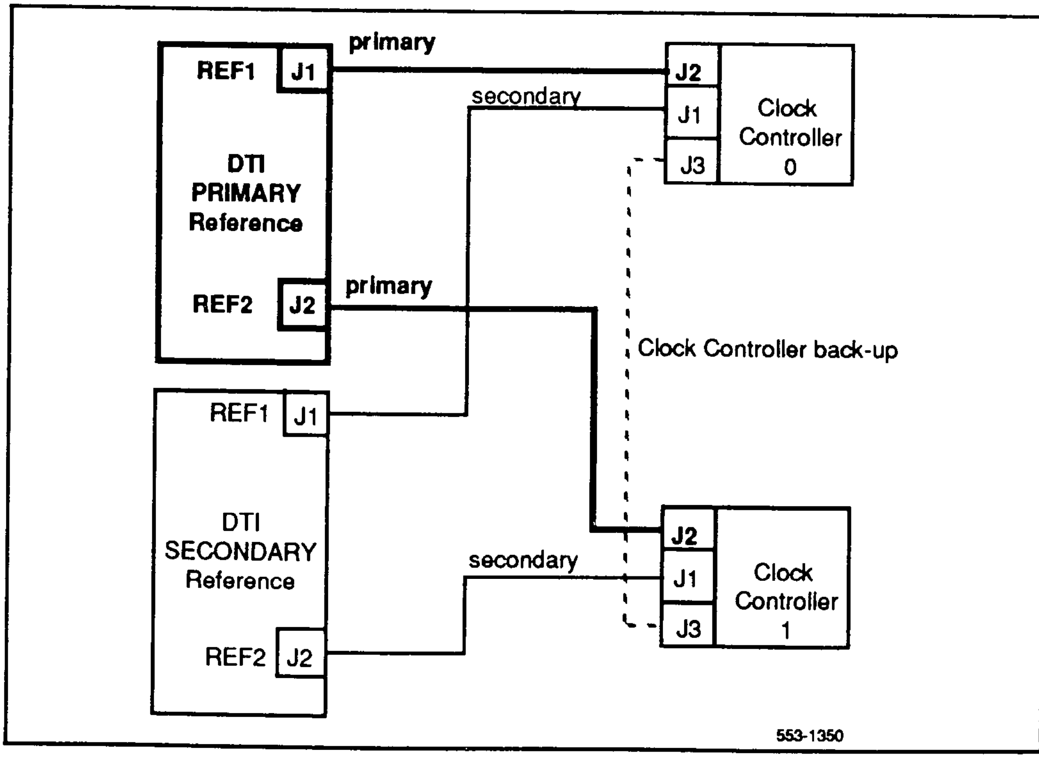
As shown in Figure 12, a Meridian 1 system with dual CPUs may have tw Clock Controllers (CC0 and CC1). One Clock Controller acts as a back-up t the other. The Clock Controllers should be completely locked to the referenc clock.

Free run (non-tracking) mode

The clock synchronization for a DTI loop may operate in free run mode if

- The loop is not defined as the primary or secondary clock reference.
- The primary and secondary references are disabled.
- The primary and secondary references are in red alarm (local alarm).

Figure 12
Clock Controller primary and secondary tracking



Reference clock errors

Meridian 1 software checks every 15 minutes to see if a Clock Controller or reference clock error has occurred.

In tracking mode, at any one time, there is one active Clock Controller which is tracking on one reference clock. If a Clock Controller error is detected, the system switches to the back-up Clock Controller, without affecting which reference clock is being tracked.

A reference clock error occurs when there is a problem with the clock driver or with the reference system clock at the far end. If the Clock Controller detects a reference clock error, the reference clocks are switched.

Automatic clock recovery

An option for automatic clock recovery can be selected in LD60 with the command EREF.

A DTI loop is disabled when it enters a red alarm (local alarm) condition. If the red alarm is cleared, the loop is enabled automatically. When the loop is enabled, clock tracking is restored in the following conditions:

- If the loop is assigned as the primary reference clock but the Clock Controller is tracking on the secondary reference or in free run mode, it is restored to tracking on primary.
- If the loop is assigned as the secondary reference clock but the Clock Controller is in free run mode, it is restored to tracking on secondary.

If the 15-minute clock check indicates the system is in free run mode:

- Tracking is restored to the primary reference clock if defined.
- If the primary reference is disabled or in red alarm (local alarm), tracking is restored to the secondary reference clock if defined.

Automatic clock switching

If the EREF option is selected in LD60, tracking on the primary or secondary reference clock is automatically switched in the following manner:

- If software is unable to track on the assigned primary reference clock, it switches to the secondary reference clock and sends appropriate DTC maintenance messages.
- If software is unable to track on the assigned secondary reference clock, it switches to free run.

Clock Controller commands (LD60)

Table 14 lists Clock Controller commands.

Table 14
Clock Controller commands (LD60)

Message	Description
DIS CC N	Disables specified system Clock Controller.
DSYL L	Disables yellow alarm (remote alarm) processing for loop L.
ENL CC N	Enables specified system Clock Controller.
ENYL L	Enables yellow alarm (remote alarm) processing for loop L.
EREF	Enables automatic switching and recovery of primary and secondary reference clocks when loops associated with these clocks are automatically enabled.
MREF	Disables automatic switching and recovery of the primary and secondary reference clocks when loops associated with these clocks are automatically disabled or in red alarm (local alarm).
SSCK N	Provides status of system clock N. Indicates the active controller as well as active primary or secondary reference clock source or free run.
SWCK	Switches the system clock from the active to the standby clock. The reference clock source remains unchanged.
TRCK xxx	Set Clock Controller tracking. Where xxx is: PCK track primary clock SCLK track secondary clock FRUN free run mode

The DTC messages indicate Clock Controller status and error conditions. They are listed in Table 15.

Table 15
DTCxxx messages

Message	Description
DTC001	Clock Controller tracking on primary source loop.
DTC002	Clock Controller tracking on secondary source loop.
DTC003	Clock Controller cannot be accessed.
DTC004	Clock Controller indicates clock aging error.
DTC005	Reference clock switched to secondary source from primary.
DTC006	Reference clock switched to free run mode from secondary or primary.
DTC007	Active reference clock is set to retrack primary.
DTC008	Active reference is free run or the Clock Controller cannot be accessed.
DTC009	Clock Controller has been switched.
DTC010	UART error is detected.
DTC011	Clock control self-test failed; error exists.
DTC012	Clock control has reference clock problem.
DTC013	Clock control has tracking problem.
DTC014	Clock control set to free run.
DTC015	Clock control set to secondary.
DTC016	Clock Controller restored from free run or secondary to tracking on primary.
DTC017	Clock Controller restored from free run to tracking on secondary.
DTC018	Cannot switch or restore to a reference clock because automatic reference clock switching option is disabled.

Replacing the Clock Controller

> **CAUTION**
> Firmly touch the metal frame of the cabinet to discharge static electricity from your body before handling cards.

Note: Do not deviate from this procedure. This procedure should not cause the system to SYSLOAD or initialize, but will stop call processing.

1 Ensure that the Clock Controller card being replaced is associated with the disabled CPU. Switch CPUs, if necessary:

LD35
SCPU

2 Disable the Clock Controller card being replaced:

LD60
DIS CC

3 On the Clock Controller card being replaced, set the ENB/DIS switch to DIS.

4 Disconnect cables from Clock Controller card being replaced.

5 Remove card from shelf.

6 Set the ENB/DIS switch to DIS on the Clock Controller card being added.

7 Ensure that the switch settings are correct.

8 Install new Clock Controller card in same slot as the defective card.

9 Reconnect cable(s) to Clock Controller faceplate.

10 Set ENB/DIS switch on new Clock Controller to ENB.

11 Enable new Clock Controller card:

LD60
ENL CC N

12 Verify normal service level; first, switch the active clock to standby:

LD60
SWCK N

If an error message results, refer to the *X11 input/output guide* (553-3001-400) for the interpretation.

Note: Switching Clock Controllers using LD60 will generate ERR20 messages. These can usually be ignored, but avoid excessive switching, especially when counters are near the maintenance or out of service thresholds. Excessive switching can generate maintenance or out of service threshold messages, or cause the DTI to be disabled. Check the counters in LD60. If necessary, reset the counters using the RCNT command.

Clock Controller switch settings

Switch settings for the QPC471 Cock Controller (vintages A, B, and C) and the QPC775 Clock Controller (currently available in Canada only) are given in Table 16.

Table 16
Clock Controller switch settings

System/Clock Controller	SW1	SW2	SW3	SW4 (1,2)	Jumper 1	Jumper 2
ST						
QPC471 A	n/a	ON	n/a	n/a	n/a	n/a
QPC471 B & C	ON	OFF	n/a	n/a	TP8-TP9	TP11-TP12
QPC775	n/a	ON	OFF	ON	n/a	n/a
N/NT single-group						
QPC471 A	n/a	ON	n/a	n/a	n/a	n/a
QPC471 B & C	ON	OFF	n/a	n/a	TP8-TP9	TP11-TP12
QPC775	n/a	ON	OFF	ON	n/a	n/a
N/NT/RT half-group						
QPC471 A	n/a	ON	n/a	n/a	n/a	n/a
QPC471 B & C	ON	OFF	n/a	n/a	TP8-TP9	TP11-TP12
QPC775	n/a	ON	OFF	ON	n/a	n/a
XN/XT						
QPC471 A	n/a	OFF	n/a	n/a	n/a	n/a
QPC471 B & C	OFF	OFF	n/a	n/a	TP8-TP9	TP11-TP12
QPC775	n/a	OFF	OFF	ON	n/a	n/a

Clock Controller cabling

The Clock Controller cabling for various system configurations is shown in Figures 13, 14, and 15.

Figure 13
Clock Controller cabling: ST and N/NT/RT half-group

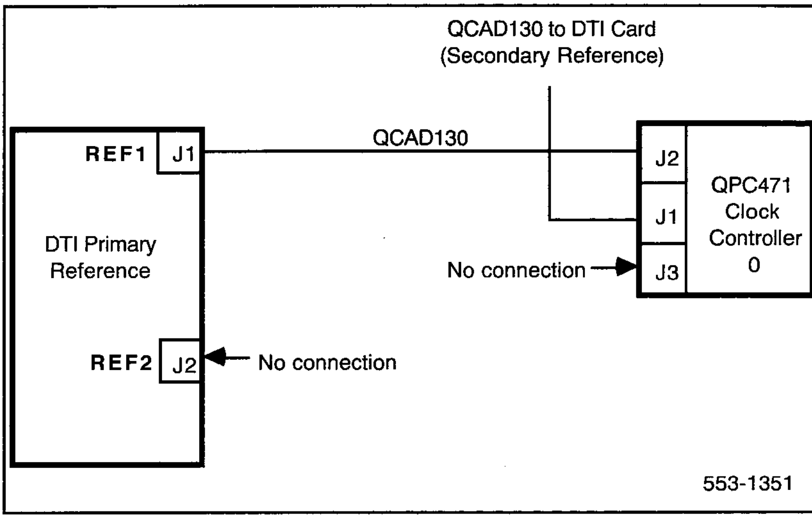


Figure 14
Clock Controller cabling: N/NT single-group

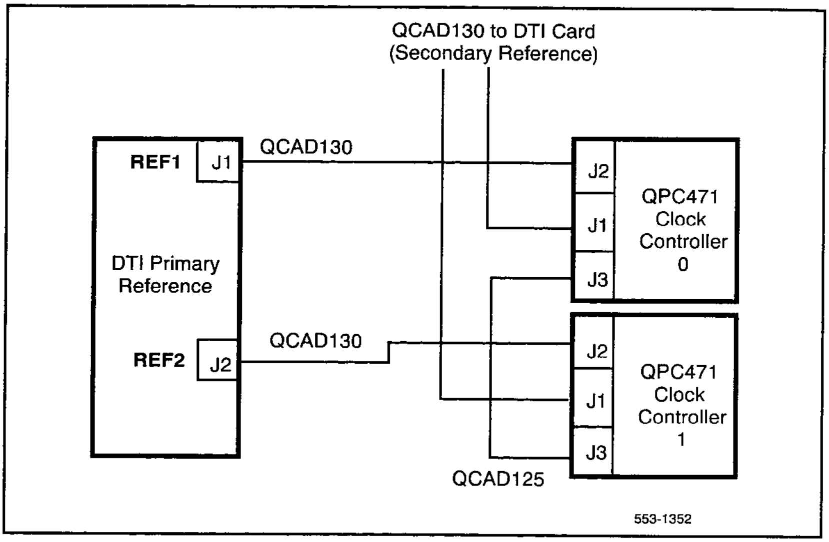
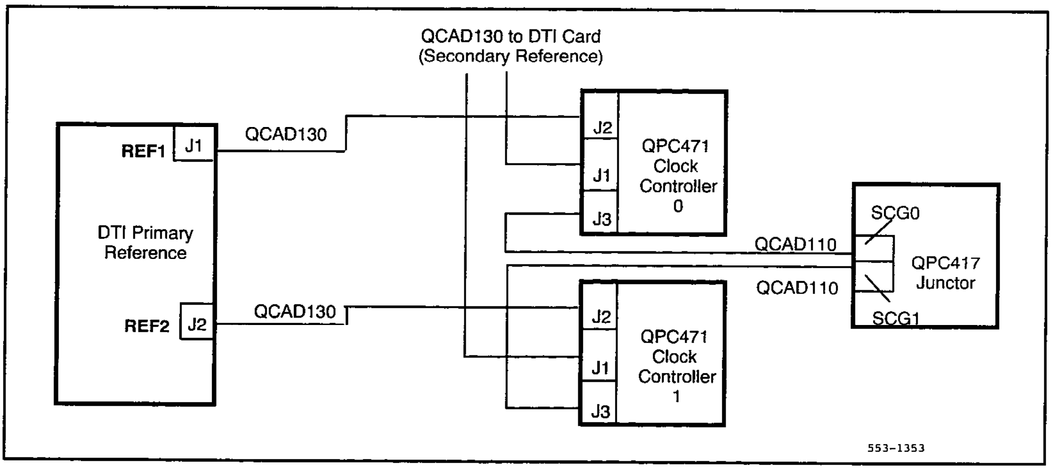


Figure 15
Clock Controller cabling: XN/XT multi-group



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Maintenance

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