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DMS-10 Family

# 600-Series Generics

## Feature and Services Description

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For Generic 602.20   Standard   August 2006

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# Contents

# Section 1: Introduction

## Scope and purpose of this publication

The purpose of this Nortel technical publication (NTP) is to introduce and describe briefly the features and services of the 600-Series DMS-10 Digital Switching System.

## Organization

Section 2 describes the services of interest to the operating company's individual subscribers. Section 3 describes features of interest to the operating company's business subscribers. Sections 4 through 11 outline the features necessary for the operation, maintenance, and administration of a DMS-10 central office. Section 12 provides an alphabetical index of the features and services available for the DMS-10 switch.

For a list of the features and hardware introduced in Generic 601, refer to the *Generic Release Summary* for Generic 601, NTP GRS-3601-601.

# Section 2: Residential services

## Introduction

This section presents general descriptions of the services of interest to subscribers served by a DMS-10 central office. Most of these services are defined in the DMS-10 software as station options. For additional information, see the NTP entitled *Data Modification Manual* (297-3601-311).

## Custom Calling Services

Custom Calling Services are selective features that allow subscribers to control the calls they make and receive. The features are activated or deactivated either by dialing special codes or by performing a hook-flash. The following paragraphs give brief descriptions of the custom calling features available on some types of lines. For station option-to-option compatibility, refer to the NTP entitled *Data Modification Manual* (297-3601-311).

### Custom Calling Services access codes

The special codes used to activate/deactivate Custom Calling Services are defined by Data Modification Orders (DMOs) as part of prefix translations. Translations can be configured for any set of codes the operating company requires, although most translations follow the standard access codes recommended by Telcordia (formerly Bellcore). For details about the translations, refer to overlay TRNS in the NTP entitled *Data Modification Manual* (297-3601-311).

### Speed Calling (SSC and LSC)

Speed Calling allows a single-party or two-party subscriber to call one of a group of numbers by dialing a one- or two-digit code. Two types of Speed Calling are available: 8-number short-list speed calling (SSC) and 30-number long-list speed calling (LSC). Each number can consist of up to 32 digits.

To assign or change an abbreviated code, the customer dials the registration code (DMO-modifiable) and waits 4 seconds to time out. A Digitone subscriber may key the registration code plus an octothorp (#) to avoid the timeout. Upon receiving the special dial tone, the subscriber dials the code number followed by the directory number it represents. When the data are received and recorded, a confirmation tone is returned to the subscriber, who then goes on-hook to complete the entry.

To make a *speed call*, the subscriber goes off-hook and, after receiving dial tone, dials the code number. A 4-s timeout is required before the DMS-10 switch processes the call. A Digitone subscriber may dial an octothorp (#) to avoid the timeout.

## Call Waiting (CWT)

The CWT feature informs a subscriber engaged in a normal talking connection that a third party is calling.

Subscriber A has the CWT feature and is in a normal connection with subscriber B. If a third subscriber (C) originates a call to A, a burst of call-waiting tone is sent to A and an audible ringing tone is sent to C. (Subscriber B hears no signaling tones.) A second burst of call-waiting tone is sent to A if he does not answer within 10 s. No further tone is sent. If A flashes the switchhook, A is connected to C and B is placed on hold. Subsequent switchhook flashes allow A to transfer between B and C.

## Cancel Call Waiting (CCWT)

The CCWT feature allows a subscriber with the CWT or UCWT option to dial an access code to cancel call waiting for the duration of the call in progress.

To activate CCWT, the subscriber dials a CCWT access code after taking the receiver off-hook. The feature can be activated prior to placing a phone call or after a talking connection has been established (for the latter, Three-Way Calling must also be assigned to the line). In either case, a confirmation tone and second dial tone are returned so that the call can then be placed normally. When the call is completed and the subscriber goes on-hook, CWT or UCWT is again operational.

## Three-Way Calling (3WC)

The 3WC feature allows a single-party subscriber to add another call to an existing two-way talking connection. This feature includes both hold and add-on capabilities.

Subscriber A has the 3WC feature and is in a normal connection with subscriber B. If A wishes to add a third party (C), A flashes the switchhook, B is placed on hold and, after receiving the special dial tone, A dials subscriber C. To bring B back into the call, A flashes the switchhook again. If, for some reason, the call to the third party is not completed, subscriber A depresses the switchhook twice to return to a normal connection with subscriber B. If, after connecting with subscriber C, subscriber A wishes to disconnect subscriber C, subscriber A depresses the switchhook for about 1 s. Upon releasing the switchhook, subscriber A will be connected only to subscriber B.

The following conditions apply to the 3WC feature:

- If a user attempts to form a three-way conference before a normal talking state is reached with the third party and no conference circuit is available, the third party is dropped and the user is reconnected to the second party.

- A user cannot receive Call Waiting indication after initiating a three-way call.
- After system initialization, three-way conference calls will not be rebuilt.
- The 3WC feature is assigned on a station basis in overlay DN. This overlay allows the assignment, deletion, and query of the 3WC feature.
- If a three-way call is in progress and equipment involved in the call, other than the NT4T03 Conference pack or the NT8T04 Network Interface pack, is removed from service, the call is disconnected.
- If a the NT4T03 Conference pack, or the NT8T04 Network Interface pack, is removed from service when a three-way call is in progress in a normal talk state, the call is switched to the mate non-busy pack; any three-way calls in progress that are not in the normal talk state are disconnected.

### Three-Way Calling enhancement

The operation of the enhanced Three-Way Calling feature complies with Bellcore requirements found in document TR-TSY-000577, *Three-Way Calling*. Prior to this enhancement, a party placed on consultation hold would be disconnected if the party initiating the three-way call disconnected before a three-way call was established. With this enhancement, the party on consultation hold will not be disconnected and the switch will ring back the party initiating the three-way call so that the call can be re-established; ring-back continues either until the party answers or until the party on consultation hold disconnects.

The following conditions apply to the 3WC feature enhancement:

- Processing of Local Coin Overtime events continues while the switch is ringing back the party initiating the three-way call.

## Three-Way Calling Operator feature

A switch-hook flash is interpreted by the DMS-10 switch as a request for either Three-Way Calling or operator recall. Switchhook-flash ambiguity is resolved by dialing an operator recall code after receiving the special dial tone.

## Call Forwarding (CFW)

The CFW feature allows a single-party subscriber to have all incoming calls forwarded to another, preselected line.

When a subscriber wishes to have calls forwarded, the subscriber dials the DMO-modifiable activation code and waits for a 4-second timeout period. This code may be up to 32 digits long. Digitone subscribers may avoid the timeout period by dialing an octothorp (#). A special dial tone is returned, and the subscriber dials the number to which calls are to be forwarded. When the station to which calls are to be forwarded answers, the feature is activated. If the station to which the call is to be forwarded does not answer or if it is busy, the subscriber hangs up and then repeats the previous steps.

If this is done within 2 minutes of the original attempt, the subscriber will hear two beeps, indicating that the CFW feature is now in effect.

To deactivate CFW, the subscriber dials the CFW deactivation code and waits for a 4-s timeout period. Digitone subscribers may avoid the timeout period by dialing an octothorp (#). Confirmation tone is returned to indicate successful deactivation.

The following restriction applies to Call Forwarding:

- In an Equal Access environment, a call cannot be forwarded to an international number.

## Fixed Destination Call Forwarding (CFF)

The Fixed Destination Call Forwarding feature allows a single-party, two-party flat-rate billing (2FR), or two-party message-rate residential billing (2MR) subscriber to have all incoming calls forwarded to another, designated DN as prearranged with the operating company. The feature eliminates the need for the subscriber to enter the forwarded-to DN when activating fixed call forwarding; entering only the fixed call forwarding activation code automatically forwards all calls to the pre-determined DN. The operating company sets up the forwarded-to DN at the time the CFF (Fixed Destination Call Forwarding) and UCFF (Usage Sensitive Fixed Destination Call Forwarding) station options are assigned to the subscriber's station.

To activate Fixed Destination Call Forwarding, the subscriber dials the DMO-modifiable activation code. When the subscriber hears a confirmation tone, fixed call forwarding is activated. To deactivate the feature, the subscriber dials the DMO-modifiable deactivation code. A confirmation tone indicates that the feature is deactivated. Activation and deactivation are allowed either from the subscriber's telephone or from a remote location.

When Fixed Destination Call Forwarding is active, the subscriber can still originate calls. When a call is forwarded, a burst of ringing is applied to the subscriber's base station to indicate that calls to the station are being forwarded. Callers receive no indication that their calls are being forwarded.

The following conditions apply to Fixed Destination Call Forwarding:

- If a subscriber attempts to activate the feature when it is already active, the subscriber is routed to overflow treatment. Multiple activations are blocked.
- If a subscriber attempts to deactivate the feature when it is already inactive, the subscriber is routed to dialing error treatment.
- The forwarded-to DN may be a maximum of 32 digits in length.

- The CFF (Fixed Destination Call Forwarding) and UCFF (Usage-sensitive Fixed Destination Call Forwarding) station options are saved through UPDT DUMP (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*).
- The designated forward-to DN is stored through Overlay CCTB (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*).

## Call Forwarding DMO Activation/Deactivation (CFWA)

The CFW DMO Activation/Deactivation feature allows operating company personnel to activate or deactivate call forwarding on a station that is assigned a call forwarding feature by Data Modification Order (DMO). This feature is especially useful in emergency situations where the base directory number is disabled or out of service and for voice mail applications in which Call Forwarding is normally activated only once.

This feature supports directory numbers that are assigned the following types of call forwarding:

- Basic Call Forwarding (CFW)
- Usage Sensitive Call Forwarding (UCFW)
- User Programmable Call Forwarding (CFB)
- Usage Sensitive User Programmable Call Forwarding (UCFB)
- User Programmable Call Forward Don't Answer (CFD)
- Usage Sensitive User Programmable Call Forward Don't Answer (UCFD)
- Fixed Destination Call Forwarding (CFF)
- Usage Sensitive Fixed Destination Call Forwarding (UCFF)

The number of rings (from 2 through 9) after which a call is forwarded can also be specified, on a station basis, for User Programmable Call Forward Don't Answer (CFD) and Usage Sensitive User Programmable Call Forward Don't Answer (UCFD).

The CFWA feature does not prevent a subscriber from changing Call Forwarding data if the appropriate access codes are defined in translations.

## Call Forwarding Limitation (CFL)

The Call Forwarding Limitation feature allows operating company personnel to limit the number of forwarded calls active at one time on a station, through assignment of the CFL station option (in overlays DN(DNCT), DN(MADN), and DN(STN)). When the threshold for the number of active call forwarding calls on the station is reached, any additional call forwarding calls placed to the station are routed to generic condition treatment (through prompt CFLR in Overlay CNFG(GCON)).

The Call Forwarding Limitation feature can operate with the following call forwarding features: CFW, CFB,CFF, CFD, SCF, UCFW, UCFB, UCFF, UCFD, and USCF.

## User programmable Call Forward Busy Don't Answer (CFBD)

User Programmable Call Forward Busy Don't Answer allows the subscriber to activate two types of call forwarding: Call Forward Busy (CFB) and Call Forward Don't Answer (CFD). CFB allows the subscriber to forward the base phone only when a busy condition is encountered. CFD allows the subscriber to forward the base phone after a specified number of rings has occurred. Both types of call forwarding are assignable station options in Overlay DN. This feature is controlled by a feature bit set by Nortel.

The subscriber may activate CFB by dialing the activation code, waiting for special dial tone, and entering the forwarded-to DN. When answer is received from the forwarded-to DN, CFB is activated. If the forwarded-to station is busy or does not answer, the activation steps can be repeated within two minutes.

To activate CFD, the subscriber dials an activation code, waits for special dial tone, then enters the number of rings desired, followed by the forwarded-to DN. The valid number of rings ranges from 2 through 9. If an invalid number is entered, the activation will fail. When answer is received from the forwarded-to DN, CFD is successfully activated. If the forwarded-to station is busy or does not answer, the activation steps can be repeated within two minutes.

To deactivate CFB or CFD, the subscriber enters the appropriate deactivation code.

The following conditions apply to the CFD feature:

- A CFD call is not forwarded to a busy station unless the station has one of the following features: Line Hunting, CFB, BTFA, BTF (if the calling party and the busy party are not members of the same IBS/EBS group), or BTFI (if the calling party and the busy party are members of the same IBS/EBS group).
- When CFBD is activated on a line that is assigned both CFBD and Don't Answer Transfer (DAT), DAT is disabled on that line.

## Remote Call Forwarding (RCFW)

A subscriber with RCFW can have calls forwarded to any directory number. Normal Direct Distance Dialing (DDD) toll charges will be applied to the Remote Call Forwarding number.

Up to 1024 directory numbers can be assigned Remote Call Forwarding Appearances (RCFA).

## Access directory numbers (ACDNs) and personal identification numbers (PINs)

Access directory numbers allow subscribers access to functions of certain features from a station. By dialing the ACDN, subscribers can enter a security code (a personal identification number) and a sequence of digits that activate, deactivate, or change the function of a feature. ACDNs are declared in the database of the DMS-10 switch that serves the customer's station.

PINs are security codes the customer dials to confirm their access to a feature. The PIN is checked against the customer's home DN, which is entered with the PIN and checked for the presence of the feature option, to determine whether or not to grant the customer access.

## Call Forward Remote Access (CFRA)

CFRA supports call forwarding activation and deactivation from a location other than the subscriber's home station.

The subscriber with CFRA dials an access DN. They then receive ringback tone, followed by special dial tone, and respond by dialing their 7-digit home DN and a Personal Identification Number (PIN). They then receive special dial tone again. The subscriber then dials the activation or deactivation number and special dial tone is returned. Finally, the subscriber dials the number to which calls should be forwarded.

When the feature is activated, the subscriber will hear two beeps, indicating that the CFRA feature is in effect. Customer deactivation requires dialing the access DN, receiving ringback tone and special dial tone, dialing the 7-digit home DN and PIN, and following the steps for deactivating CFW.

The following restriction applies to CFRA:

- Except for SIP subscribers, the calling party's station must subscribe to Digitone/touch tone (DGT) service in order to perform CFRA. For SIP lines, DGT is not compatible. To use CFRA, these devices must be capable of generating DTMF tones. Calls terminating to the DMS-10 over a SIP trunk will be supported providing the calling party’s station is capable of generating DTMF tones.

## Call Forward Internet Down (CFID)

For more information see "Call Forward Internet Down (CFID)" on page 8-4.

## Remote Access to Simultaneous Ringing Service

Simultaneous ringing (SRNG) supports service activation, deactivation and list editing from a location other than the subscriber's home station. The subscriber with SRNG dials an access DN. They then receive ringback tone, followed by special dial tone and respond by dialing their 7-digit home DN and a Personal Identification Number (PIN). After a 4 second timeout period or after the subscriber enters an octothorp (#), special dial tone is returned. The subscriber then dials the SRNG service access code to access the Screen List Editing (SLE) functionality.

The following restriction applies to SRNG:

- The calling party's station must subscribe to Digitone/touch tone (DGT) service in order to perform SRNG.

## Residential User Transfer (UTF)

A subscriber with Residential User Transfer can transfer an established call to another line by performing a switchhook flash, receiving special dial tone, dialing the third party, and performing a disconnect at any time following completion of dialing. 3WC or Usage-Sensitive 3WC (U3WC) must be configured on stations with Residential User Transfer. The Residential User Transfer subscriber will be billed for all calls transferred by that subscriber and for charges incurred after the Residential User Transfer subscriber leaves the connection. The Residential User Transfer subscriber can transfer any call to any other station except when the resulting connection would be between outgoing trunks.

## Ring Again (RAG)

The RAG feature allows a subscriber who reaches a busy station to receive a special ring indicating that a previously busy line has become idle within a prescribed time period.

To activate RAG, subscriber A originates a call to Subscriber B. If subscriber B is busy, subscriber A then performs a switchhook flash, receives special dial tone, dials the RAG activation code, receives a confirmation tone, then hangs up. At this point subscriber A is free to make and receive calls normally. When subscriber A and subscriber B become idle within a prescribed time period, subscriber A's station re-rings. Subscriber A answers and the DMS-10 switch processes the call to subscriber B.

RAG is monitored by two system timers. The *ring again queue timer*, administrable in one minute increments between five and 30 minutes, measures the period that a RAG request remains queued to the target station. The *ring again recall timer*, which is administrable in six second increments between 12 and 30 seconds, determines the length of the period that the originator of a RAG request receives RAG re-ring.

RAG can be deactivated either manually or automatically. To manually deactivate RAG, subscriber A goes off-hook and dials the RAG deactivation code. The RAG request is removed and the Ring Again Queue Timer is canceled. After RAG is deactivated, a confirmation tone is received by subscriber A. The RAG request is automatically deactivated if either the RAG request timer expires or the Ring Again recall timer expires.

The following conditions apply to the Ring Again feature:

- The originator and target must be served by the same switch.
- The originating station must have the RAG station option and the target station must not have the RAG denied station option.
- The originating station can only have one RAG request active at any time.
- The target station cannot exceed 16 RAG requests, which are served in the order in which they are received.
- Multiparty lines are not allowed to originate RAG requests.
- RAG cannot be activated by a station user who is already in a 3WC.
- CWT has priority over RAG when both features are applicable.
- When a target has any type of CFW active, RAG requests will be denied against the target.

## Ring Again Denied (RAGD)

A subscriber with RAGD can deny other subscribers the ability of invoking RAG requests against that station.

## Usage Sensitive Three-Way Calling (U3WC)

A subscriber with U3WC can add another party to an existing two-way talking connection by flashing switchhook, dialing a third party, then flashing switchhook. The U3WC subscriber that adds the third party to the connection is billed for each successful addition of a third party to an existing two-way connection. If a billing register is not available, the request may be allowed (with no billing record generated) or prevented, at the option of the operating company.

The service provider can determine whether access to the feature should be through a service access code or through hook-flashing.

### Office-wide Usage Sensitive Three-Way Calling (O3WC)

O3WC extends Usage Sensitive Three-Way Calling to subscribers on an office-wide basis. Thus, the feature enables single-party subscribers in the office who do not already have flat-rate billing for Three-Way Calling to be billed on a per-use basis. The feature also enables the service provider to deny the feature to individual stations in the office that do not want a bill accumulated for three-way calling. In addition, the service provider can determine whether access to the feature should be through a service access code or through hook-flashing.

When O3WC is active in an office, the User Transfer (UTF) feature can also be added to single-party subscribers' stations.

## Usage Sensitive Call Forwarding (UCFW)

The UCFW feature allows a subscriber to forward all incoming calls to a preselected destination, as is performed with CFW, using activation and deactivation that are different from those used for CFW. Because a billing record is generated both at the time of activation and at the time of deactivation, the subscriber may be billed either for each UCFW activation or for the duration of the UCFW activation. This billing is separate from any toll charges that apply if the call from the subscriber's station to the forwarded destination is a toll call. If no billing register is available when UCFW is deactivated, deactivation is allowed but no billing record is generated. If no billing register is available when UCFW is requested, the request may be allowed (with no billing record generated but with the activation still recorded in the IDS as above) or prevented, at the option of the operating company.

## Usage Sensitive User Programmable Call Forward Busy Don't Answer (UCBD)

Usage Sensitive User Programmable Call Forward Busy Don't Answer is similar to User Programmable Call Forward Busy Don't Answer, except in billing. In general, usage sensitive billing records are generated both at the time of activation and at the time of deactivation. Therefore, the subscriber may be billed either for each activation or for the duration of the activation. However, if a CFW type is activated or deactivated through DMO, no billing record is generated.

Usage Sensitive Call Forward Busy (UCFB) and Call Forward Busy (CFB) are not compatible station options. Also, Usage Sensitive Call Forward Don't Answer (UCFD) and Call Forward Don't Answer (CFD) are not compatible station options.

## Usage Sensitive Call Waiting (UCWT)

The UCWT feature provides notification to the subscriber engaged in a talking connection that a third party is calling, as is performed by CWT. The UCWT subscriber is billed only once, regardless of the number of times an original connection is suspended and a single third party is re-connected. However, the UCWT subscriber is billed for each time the original connection is suspended and a new third party is connected. If a billing register is not available, the call may be allowed (with no billing record generated) or prevented, at the option of the operating company.

## ISDN Call Forwarding (CF)

ISDN call forwarding features are assigned in Overlay DN through the DNCT prompting sequence. CF enables subscribers to redirect calls from one directory number (DN) to another DN. CF consists of the following four subfeatures:

- Call Forwarding All (CFW)
- Call Forwarding Busy (CFB)
- Call Forwarding Fixed Destination (CFF)
- Call Forwarding Don't Answer (CFD)

The CFW feature redirects all incoming calls forwarded to a subscriber programmed remote directory number (RDN), regardless of the line's busy or idle status. CFB redirects an incoming call only when a busy condition is encountered. CFF redirects all incoming calls to a designated DN as prearranged with the operating company. CFD redirects the call after a specified number of rings has occurred. When activated by operating company personnel, CFD always provides two rings before forwarding occur. Subscribers have the option of activating CFD to provide two, up to nine, rings before forwarding occurs. The number of rings may vary for ISDN subscriber terminals that are not compliant with the audible ringing tone (2 seconds on, 2 seconds off) defined in SR 2661.

For VI call types, a courtesy call is made when an ISDN subscriber attempts to activate a CF feature, and is activated when the RDN translates successfully. CF subfeatures are assigned to Directory Number Call Types (DNCT) for either VI or CMD call types. For the CFW subfeature, a Feature Activator (FA) and/or Feature Indicator (FI) may be assigned through overlay ISDN, prompting sequence TCGN. For non call associated VI and CMD activations, ISDN subscribers can activate or deactivate the CF feature outside the context of a call by using a keystroke or similar action through a Feature Activator. The FI provides a visual indication of the CFW status on the subscriber's ISDN terminal. All other methods of activating and deactivating CF subfeatures for an ISDN terminal are identical to the process used to activate or deactivate a Meridian Business Set (MBS).

## ISDN Call Reference Busy Limitation (CRBL)

Call Reference Busy Limit, CRBL n(n), where n(n) is a value from 1 to 12. Indicates the total number of simultaneous call references the DMS-10 allows to be active for a DNCT.

## ISDN Hold Capability (IHC)

IHC allows ISDN users on an established speech, 3.1 kHz audio, or circuit-mode data call to place that call on hold; and to later retrieve the call from hold. While a call is on hold, a subscriber is able to answer or originate other calls. A call may remain on hold for an unlimited time period and an ISDN terminal can have multiple calls on hold at the same time. IHC also serves as a common function for ISDN Flexible Calling and ISDN Additional Call Offering which require hold capability for multiple call management. IHC is assigned to an ISDN terminal through the Terminal Service Profile (TSP) and is available to all DNCTs on the terminal. IHC functions in a similar manner to the MBS Call Hold feature.

Placing a call on hold releases the call's original B-channel connection. Once released, the B-channel is available to receive or originate additional calls; or to retrieve the call on Hold. If a subscriber is assigned the B-channel Reservation option, a B-channel is reserved for use by the holding ISDN terminal. A B-channel continues to be reserved as long as at least one call is on Hold, and there are no active calls. To ensure that at least one B-channel is always available, an active call on one B-channel releases the other B-channel from reserved status.

A subscriber can place an active call on hold by requesting Hold activation, which holds the call and removes the B-channel connection. The subscriber is left with three options, retrieve the call on Hold, initiate a new call on an idle DN appearance, or retrieve another (previous) call on Hold. Hold cannot be invoked on an E911 call, including calls that conference an E911 call.

Hold/Retrieve Notification allows a party on Hold (remote party) to receive notification that Hold was invoked. When the call is retrieved, the remote party receives notification that Retrieve was invoked. In a case where the ISDN subscriber invokes Hold before the remote party answers the call, notification is provided to the remote party when the call is answered. If Hold and Retrieve are both invoked before the remote party answers the call, no notification is provided. Hold/Retrieve Notification is available for inter-switch calls through SS7 facilities that do not encounter interworking.

## ISDN Flexible Calling (FC)

FC is a set of capabilities that allow an ISDN subscriber to establish and control two or more concurrent calls. In addition, FC capabilities include the ability to build conferences, drop the last party added to a conference and to transfer conferenced (bridged) or non-conferenced calls. Circuit mode data calls cannot be bridged through FC. FC is assigned to an ISDN terminal through the TSP and is available to all DNCTs on the terminal. The ISDN Hold Capability (IHC) feature works with FC to allow call and conference hold and retrieval, therefore IHC is automatically assigned to terminals assigned with FC. This feature functions in a similar manner to MBS Three-way Conference with Consultation Hold (3WC) and Call Transfer (UTF) features.

FC uses ISDN Hold Capability (IHC) to hold and retrieve individual calls and conferences. An ISDN Retrieve request allows the bridging of calls to form a conference. FC supports a maximum three-party conference size. Conference chaining is allowed, including mixed chains of FC conferences and 3WC conferences.

Flexible Calling requires establishing an FC controller by issuing a conference Feature Activator from an ISDN terminal. The controller can be established during call origination, dialing, while receiving Ringback Tone or after the call has been answered. A conference Feature Indicator is sent to the controller's terminal providing a visual indication that conference has been activated. Because FC allows only one controller, an ISDN terminal is limited to a single conference at any one time. A conference feature indicator remains active until the conference is idled. In this context, the FC controller refers to the DNCT appearance where a Feature Indicator displays *CONFERENCE*. The term *conference* refers to a bridged call that can consist of up to three parties.

## ISDN Additional Call Offering (ACO)

ACO allows calls to be presented to an ISDN subscriber's terminal in situations where the called DNCT is Interface Busy and the calling subscriber would normally receive Busy treatment. ACO consists of two originating features and two terminating features. ACO functions in a similar manner to the MBS Call Waiting feature.

As a requirement for ACO feature operation, an ISDN terminal must also be assigned ISDN Hold Capability (IHC). This is because ACO calls do not have a specified B-channel, and IHC provides a method of releasing a B-channel for the ACO “waiting” call.

When an incoming call is directed at a DNCT that is Interface Busy, the calling party receives Ringback tone and the call is presented, without a specified B-channel, to an idle appearance on the called party's DNCT. The called party must use IHC to manually hold an active call, and free a B-channel for the ACO “waiting” call.

The two originating ACO features are Call Waiting Originating (CWTO) and Dial Call Waiting (DCWT). CWTO and DCWT are described as part of the MBS Call Waiting feature, and may only be assigned to ISDN directory numbers with voiceband call types.

The two terminating ACO features are Unrestricted and Restricted. Unrestricted ACO is invoked against a called ISDN party, however the calling party can be either an interswitch or non-ISDN call. Restricted ACO functions in an identical manner to Unrestricted ACO, but only for EBS group calls where either the CWTO or DCWT priority features have been invoked by the calling party.

## ISDN Multipoint EOC

The ISDN Multipoint EOC feature, introduced in Generic 412.20, enhances Layer 1 performance monitoring and maintenance capabilities for IDC ISDN NTBX27 lines by enabling the DMS-10 switch to communicate with and test the intermediate line units through an expanded MP-EOC command/response message set.

A mp-eoc configuration occurs on a ISDN BRI U-loop when intermediate Line Units exist between the BX27 linecard and the NT1. One to six intermediate line units can be used to extend the existing U-loop range beyond 18K feet (per unit) from the IDC.

The following MP-EOC performance monitoring (PM) information is collected for each intermediate line unit:

- block errors - the number of blocks (superframes) in which a CRC violation occurred
- errored seconds - the occurrence of one or more block errors in a single direction of transmission during a one-second interval
- severely-errored seconds - the occurrence of three or more block errors in a single direction of transmission during a one-second interval

The following counts and history data are maintained in each intermediate line unit:

- current hour (block errors, errored seconds, severely-errored seconds)
- current day (errored seconds, severely-errored seconds)
- previous hour (block errors, errored seconds, severely-errored seconds)
- previous day (errored seconds, severely-errored seconds)
- hourly history, for the seven most-recent hours (errored seconds)

Thresholds for these error types are established for each intermediate line unit and are used to determine whether the ISDN line is operating within prescribed tolerances. When a threshold is exceeded, MP-EOC PM causes the IDC to send a message describing the condition to the DMS-10 switch.

In addition to providing the DMS-10 switch the ability to test and monitor intermediate units, MP-EOC PM also provides the DMS-10 switch the capability to query the status of the intermediate line units to determine current and history counts for the unit's error registers, to determine the unit's threshold values, and to determine whether alarm mode (that is, the ability of the unit to report a threshold violation) is enabled for the unit.

## Long Distance Alert

The Long Distance Alert (LDA) feature causes distinctive ringing to be applied to an idle station, or causes distinctive call waiting tones to be applied to a busy station, when a call terminating on the station is a long distance call. Because subscribers generally place a high priority on retrieving incoming long distance calls, the purpose for the feature is to enable subscribers to distinguish between long distance and local calls in order to promote increased completion of long distance calls.

The Long Distance Alert feature recognizes toll calls originating either from a trunk (both in-band and ISUP) or from a line. When a station assigned the LDA option is idle and receives a long distance call, a distinctive ringing pattern is applied to the station. If the station is busy in a regular two-party call when a long distance call is placed to it, a distinctive call waiting tone is applied, and the incoming long distance call is held. The station can then connect to the waiting call either by hook-flashing to put the existing call on hold or by disconnecting from the existing call. If, however, the station is busy in a call configuration other than a regular two-party call, the station is not provided with a call waiting tone and busy tone is applied to the long distance calling party's station.

A subscriber assigned the LDA option is able to activate or deactivate the Long Distance Alert feature by entering codes defined by the operating company. A subscriber who successfully activates or deactivates LDA receives a confirmation tone. Unsuccessful activation or deactivation, or attempts to activate or deactivate LDA from a station not assigned the option, results in standard dialing error treatment.

The Long Distance Alert (LDA) station option can be assigned only to LCM, RSC-S, IDT, RCU, and SLC-96 non-ISDN, single-party lines that are not members of an EBS/IBS group. The station option cannot be assigned to PE lines, to PRI interfaces, or to BRI lines. The feature is available to offices configured for any ringing type.

The LDA feature provides the operating company the option of specifying (through Overlay TG (INC) or (2WAY)) that all incoming calls over a designated in-band or ISUP trunk group will be considered long distance calls. The operating company can also specify (through Overlay PRI (PRI)) that all calls originating on a designated incoming or 2-way line trunk group will be considered long distance calls.

### LDA ringing parameters

The LDA feature employs the ringing patterns for different equipment types shown in Table 2-A and ringing cadences shown in Table 2-B.

**Table 2-A:**
**Long Distance Alert ringing patterns**

| Line Equipment Type | Coded Ringing | Multifrequency/ Multifrequency-Bell Ringing | Superimposed Ringing |
|---|---|---|---|
| LCM/RSC-S | L - S - S | S - S | L - S |
| SLC-96 | L - S - S | S - S | L - S |
| RCU | L - S - S | S - S | L - S |
| IDT | L - S - S | S - S | L - S |

**Table 2-B:**
**Long Distance Alert ringing cadences**

| LCM - based Equipment Ringing | Coded Ringing | Multifrequency Ringing | Multifrequency-Bell Ringing | Superimposed Ringing |
|---|---|---|---|---|
| Ringing | 1500 ms | 500 ms | 500 ms | 1500 ms |
| Silent | 500 ms | 500 ms | 500 ms | 500 ms |
| Ringing | 500 ms | 500 ms | 500 ms | 500 ms |
| Silent | 500 ms | 4050 ms | 4500 ms | 4000 ms |
| Ringing | 500 ms | | | |
| Silent | 2500 ms | | | |

# Custom Local Area Signaling Services (CLASS)

CLASS is a feature set that provides subscribers greater convenience in placing calls and greater control over incoming calls. The CLASS features may be offered on either a usage-sensitive or subscription basis, but are available only for single line service. The features are operational on all DMS-10 remote switches except the DMS-1R: the DMS-1R does not support the Calling Number Delivery (CND), Calling Name Delivery (CNAM), or Calling Identity Delivery and Suppression (CIDS) features. CND is supported on the SLC-96. Additional, non-DMS-10 switch hardware required for the operation of CLASS features includes a vendor-supplied interactive announcement system. Although CLASS features function in an intra-office environment, for inter-office application the DMS-10 switch must be part of a CCS7 network.

The following paragraphs describe CLASS feature access and each of the CLASS features. For information concerning station option-to-option compatibility, refer to the NTP entitled *Data Modification Manual* (297-3601-311).

## Custom Local Area Signaling Services access

Each of the CLASS features is activated and deactivated by dialing a special access code defined by the operating company. The special codes are defined by Data Modification Orders (DMOs) as part of prefix translations. Translations can be configured for any set of codes the operating company requires. For more information about translations, refer to Overlay TRNS in the NTP entitled *Data Modification Manual* (297-3601-311).

### Hookflash access

A subscriber currently in talking state may hookflash and then activate or deactivate any CLASS feature except CNB. For stations assigned CLASS features, but not assigned the three-way calling (3WC) feature, the subscriber hookflashes and receives a special dial tone. The subscriber may then dial a CLASS feature access code. A subsequent hookflash ends the CLASS session and returns the subscriber to the original call. For stations assigned CLASS features and the 3WC feature, the subscriber hookflashes and, after receiving a special dial tone, may then either access a CLASS feature or make another phone call. If the subscriber accesses a CLASS feature, a subsequent hookflash ends the CLASS session and returns the subscriber to the original call. Single-party stations not assigned any CLASS features, in an office with office-wide Customer Originated Trace (COT), may hookflash for access to COT.

The Local Coin Overtime / Custom Calling Interface feature enables parties to utilize the CLASS features that require the use of hookflashes when called from a coin phone. For additional information about the LCO feature, refer to Section 5 of this NTP.

## Anonymous Call Rejection (ACR)

ACR enables CLASS subscribers to reject anonymous calls made to their stations. Such calls do not terminate at the station but instead are routed to a tone or to an announcement. The called party receives no alerting that such a call has been rejected.

Two different methods used in the DMS-10 switch to determine call anonymity are available to an operating company. The method selected by the operating company applies to an entire office. One method determines that a call is anonymous if the calling number is restricted from being delivered; this method complies with Bellcore's Anonymous Call Rejection feature design. The other method, which complies with NTI's DMS-100 family Anonymous Call Rejection feature design, determines that a call is anonymous if all displayable information is restricted, depending upon the capabilities of the called party.

Regardless of the method chosen to determine call anonymity, ACR applies only to calls for which display information has been intentionally blocked by the calling party or is restricted from delivery; if either office in an interoffice call does not have access to the CCS7 network, a call is not considered anonymous because calling party information cannot be supplied. Calling number delivery can be restricted or blocked for the following reasons:

- The calling party has a *private* DN and doesn't invoke a number delivery service (Calling Number Delivery [CND] or Calling Identity Delivery Suppression [CIDS] delivery) in setting up the call.
- The calling party has a *public* DN and invokes a number blocking service (CNAB or CIDS) in setting up the call.
- In an intra-office call, the office is set up to suppress calling number information.
- The originating office in an inter-office call is set up to suppress calling number information.
- The terminating office in an inter-office call is set up to suppress calling number information.
- Calling Forward Privacy feature.
- Calling name and number display suppression is in effect for the Centrex group for intra-group calls (see prompt SNND, Overlay HUNT (EBS), NTP 297-3601-311, *Data Modification Manual*).

Calling Name Delivery can be restricted or blocked for the following reasons:

- The calling party has a *private* name and doesn't invoke a name delivery service (Calling Name Delivery [CNAM] or CIDS delivery) in setting up the call.
- The calling party has a *public* name and invokes a name blocking service (CNAB or CIDS suppression) in setting up the call.
- In an intra-office call, the office is set up to suppress calling name information.
- The originating office in an inter-office call is set up to suppress calling name information.
- The terminating office in an inter-office call is set up to suppress calling name information.
- The terminating office in an inter-office call is set up to automatically change the name display status to *private* if the number display status is *private*.
- Calling Forward Privacy feature.
- Calling name and number display suppression is in effect for the Centrex group for intra-group calls (see prompt SNND, Overlay HUNT (EBS), NTP 297-3601-311, *Data Modification Manual*).
- Forced name privacy is in effect for the Centrex group for intra-group calls forwarded outside of the Centrex group (see prompt FNPR, Overlay HUNT (EBS), NTP 297-3601-311, *Data Modification Manual*).

To activate or deactivate ACR, the subscriber dials a designated service access code. If ACR cannot be activated due to the lack of an essential resource (for example, Auxiliary Data Store), the call is routed to the office's CLASS generic treatment. When ACR is activated, the subscriber receives either a confirmation announcement or tone.

When a caller places an anonymous call to a station that has ACR activated, the caller receives either a tone or an announcement. If the caller remains on the line for more than ten seconds after receiving the ACR response, the caller receives normal time-out treatment.

ACR is available to single-party lines on a flat-rate billing basis, on a usage-sensitive billing basis (UACR), and on an office-wide, usage-sensitive billing basis (OACR). ACR and UACR may be assigned to a line that either has or doesn't have display features assigned to it. The Deny ACR (DACR) station option can be assigned to individual stations in an office configured for OACR to deny ACR service to only those lines.

The following conditions apply to ACR:

- If no display feature is assigned to the line, incoming calls will be rejected, based upon number only.
- The following are intra-office ACR feature limitations:
  - If the office is set up to determine anonymity based upon calling number presentation only and has OSUP and/or TSUP enabled (prompts OSUP and TSUP in the SYS prompting sequence of Overlay CNFG), a call will never be allowed to terminate to a subscriber's line that has ACR active.
  - If the office has OSUP and/or TSUP enabled, and ONAS and/or TNAS enabled (prompts ONAS and TNAS in the SYS prompting sequence of Overlay CNFG), a call will never be allowed to terminate to a subscriber's line that has ACR active, regardless of the type of ACR in use.
  - If the office has OSUP and/or TSUP enabled, and BQPN enabled (prompt BQPN in the DISP prompting sequence of Overlay CNFG), a call will terminate to a subscriber's line that has ACR active only if anonymity is based upon calling name and number, a name display option is assigned to the terminating line, the calling party invoked CNAB service in setting up the call, and the calling party has a private name in the Service Control Point (SCP) calling name database.
- The following are inter-office ACR feature limitations:
  - When the terminating office is set up to determine anonymity based upon calling number presentation only, an incoming call from an office with office-wide number suppression enabled will never be allowed to terminate to a subscriber's line that has ACR active.
  - An incoming call from an office with office-wide name and number suppression enabled will never be allowed to terminate to a subscriber's line that has ACR active, regardless of the type of ACR in use.
  - When the terminating office has TSUP enabled, an incoming call originating from an office with office-wide name suppression enabled will never be allowed to terminate to a subscriber's line that has ACR active, regardless of the type of ACR in use.
  - When the terminating office has TSUP and TNAS enabled, an incoming call from any office will never be allowed to terminate to a subscriber's line that has ACR active, regardless of the type of ACR in use.
  - When the terminating office has TSUP enabled and is set up to determine anonymity based upon calling number presentation only, an incoming call will never be allowed to terminate to a subscriber's line that has ACR active.

- When the terminating office has TNAS enabled, an incoming call originating from an office with office-wide number suppression enabled will never be allowed to terminate to a subscriber's line that has ACR active, regardless of the type of ACR in use.
- When the terminating office has BQPN enabled, an incoming call originating from an office with office-wide number suppression enabled will terminate to a subscriber's line that has ACR active only if anonymity is based upon calling name and number, a name display option is assigned to the terminating line, an ISUP GN parameter was received containing calling name information and allowing calling name presentation, and Use Name Characters in ISUP GN (UNCH) is enabled.
- An incoming call to a terminating office with both TSUP and BQPN enabled will terminate to an active ACR line only if anonymity is based upon calling name and number, a name display option is assigned to the terminating line, an ISUP GN parameter was received containing calling name information and allowing calling name presentation, and Use Name Characters in ISUP GN (UNCH) is enabled.
- A call will terminate to an active ACR line where anonymity is based on name only if the default calling name status is set to *public* in the calling name database and the calling number is public. An incoming call from another office over a SIP trunk will set the calling name status to *private* when the calling number status is *private* due to a limitation in the SIP protocol. When the calling number is public the calling name status is determined by the default status in the calling name database. The SIP protocol does not allow the passing of CNAB information in regards to calling name display.

### Interaction of Anonymous Call Rejection with other features

**Automatic Callback / Automatic Recall** If a subscriber whose station is assigned ACR and either Automatic Recall (AR) or Automatic Callback (ACB) tries to call a number that is marked *private*, then ringback associated with the ACB/AR request is not blocked. This is because the terminating office determines call anonymity only when the called station receives a call.

**Call Forwarding features** The following conditions apply to Call Forwarding features and Anonymous Call Rejection:

- Anonymous calls terminating at a station with ACR and CFW (Call Forwarding), CFB (Call Forwarding Busy), or CFD (User-Programmable Call Forwarding Don't Answer) active are routed to ACR generic treatment.
- Calls terminating to a station with both ACR and SCF (Selective Call Forwarding) active are forwarded if the calling DN is present on the screening list, irrespective of the calling name and number display status.

- The originating party's name and number display status are used rather than the forwarding station's name and number display status to determine whether a call is anonymous.
- A subscriber with both ACR and a call forwarding feature cannot dial the Call Forwarding service access code followed by the ACR activation/ deactivation code.
- Calling name and number display suppression is in effect for the Centrex group for intra-group calls (see prompt SNND, Overlay HUNT (EBS), NTP 297-3601-311, *Data Modification Manual*)

**Call Waiting features** A subscriber with both Call Waiting (CWT) and ACR doesn't receive CWT tones for incoming anonymous calls; the incoming anonymous call is rejected. A subscriber with ACR and Cancel Call Waiting (CCWT) or Dial Call Waiting (DCWT) may not dial the CCWT service access code followed by the ACR activation code ACR.

**Directed Call Pickup features** A subscriber with ACR and Directed Call Pickup (DCPU) or Directed Call Pickup From Any station (DPUA) may not dial the DCPU/ DPUA service access code followed by the ACR activation code.

**CLASS display blocking features** A subscriber with ACR and Calling Name Delivery Blocking (CNAB), Calling Number Delivery Blocking (CNB), Calling Number Delivery Suppression (SUPR), or Calling Identity Delivery and Suppression (CIDS) may not dial the service access code for one of the features followed by the ACR activation code.

**Directory number hunt groups** An anonymous call that terminates at the dialed DN of a hunt group member that has ACR active is routed to ACR generic treatment. If, however, an incoming anonymous call *hunts* to that station, the DMS-10 switch attempts to complete the call.

**Three-Way Calling** When a subscriber whose station is assigned 3WC attempts to include a third party who has ACR active, the third party's office checks the presentation status of the call. If the calling name or number associated with the call is restricted from display, the third party is not included in the call.

**User Transfer (residential and IBS)** When a subscriber attempts to transfer a call to another party who has ACR active, the second party's office checks the presentation status of the call. If the calling name or number associated with the call is restricted from display, the call is not transferred.

## Automatic Callback (ACB)

ACB causes the switch to automatically place a call to the last DN that the subscriber dialed. When the subscriber dials the ACB activation code, the DMS-10 switch retrieves the last DN dialed by the subscriber (except DNs for operator-assisted calls, directory-assistance calls, and access codes) and places a call to that station if it is idle.

If the station is busy, the switch monitors the busy/idle status of the calling party's station and of the called party's station for a specified time period (5 to 30 minutes). Scanning for a change in status of the DN to be called back is performed at the terminating switch if possible, and if not possible, at the originating office through the sending of periodic TCAP requests to the terminating switch. When both stations are idle, the switch applies distinctive ringing to the calling party's line. If the calling party's station is also assigned the calling number delivery (CND) option, the number of the station to be called displays. If the calling party goes off-hook in response to the distinctive ringing, the switch attempts to place the call.

Up to 30 ACB requests can be queued for each subscriber. A request is removed from the queue either when the switch places a call to the called party or when the monitoring period for that request expires. The subscriber can cancel all ACB requests by dialing a special deactivation code. The subscriber can continue to originate and receive calls without affecting the queued requests.

The following conditions apply to ACB:

- Although the DMS-10 switch allows up to 30 requests in each station's originating queue, only one request, either ACB or AR, is allowed for a single destination directory number. In addition, only 30 requests are allowed in each station's originating queue for ACB and AR combined.
- Line status checks and queuing are performed only on the called station and not on the directory number hunt (DNH) group to which the station belongs.

### Automatic Callback for ISDN subscribers

The Automatic Callback feature is available to ISDN subscribers. The feature is activated and deactivated by the subscriber either by dialing ACB activation and deactivation codes or by operating a feature activator (single keystroke or other similar action) at an ISDN terminal. There are two types of feature activator: toggle and paired. The toggle feature activator enables one feature activator to be used to activate and deactivate (toggle) ACB. The paired feature activator provides two feature activators: one feature activator is used to activate ACB while the other feature activator is used to deactivate all requests made by its paired feature activator. ISDN terminals also support a visual feature indicator for ACB feature activators, lighting a lamp associated with the service.

**ISDN ACB call processing** When an ACB call is placed to a DN to be called back and the DN is not busy, the call is processed immediately. When the call is placed to a DN that is busy, the request is queued and the DN is monitored for a predetermined period of time (5 to 30 minutes). Up to 30 ACB requests from each ISDN interface can be queued, regardless of the number of terminals assigned to the interface. When the targeted DN becomes idle, the calling party is notified, even if the party is busy, and the target DN displays on the calling party's terminal. If the calling party goes offhook in response to the notification, the call is placed.

The following conditions apply to ISDN ACB:

- The target DN is always displayed during idle notification, whether or not the ACB subscriber's terminal is assigned the CND option.
- Dialing an ACB deactivation code cancels only the requests made using the activation code; feature activator requests are not affected.

### Office-wide Automatic Callback (OACB)

OACB extends Automatic Callback to subscribers on an office-wide basis. Thus, subscribers do not have to subscribe to the ACB feature in order to use it. The feature also provides subscribers with convenient ACB access by providing an announcement that instructs the subscriber to dial “1" in order to activate ACB when the line being called is busy. Subscribers can also activate ACB by hook-flashing and dialing an access code (*66).

Single-party subscribers in the office who do not already have flat-rate billing for Automatic Callback are billed for Automatic Callback on a per-use basis. Flat-rate ACB subscribers who use the OACB feature are billed on a flat-rate basis.

When the OACB feature is installed in an office, the feature can still be denied for individual subscribers. In addition, the OACB prompting announcement can also be suppressed for individual stations in the office.

The following conditions pertain the OACB feature:

- Billing for OACB is the same as that for UACB. Flat-rate ACB and UACB billing takes priority over OACB billing.
- For inter-office calls, the OACB announcement is performed only if an ISDN release message (REL) with the cause, “user busy,” and the status indicator, “ISUP used all the way” is received.
- OACB announcements must be supplied by the operating company. Different announcements must be supplied for flat-rate OACB subscribers and for usage-sensitive OACB subscribers. The announcement numbers are configured in Overlay CNFG (CLAS), prompts ACB# (flat-rate) and UCB# (usage-sensitive) (see *Data Modification Orders*, 297-3601-311).
- The OACB announcement is not provided for calls terminating to multiparty lines, MF trunks, to Line trunks, or to coin phones and is also not provided for N11 calls, for 800/888/900 calls, or for directory assistance calls.
- The OACB announcement is not provided when the called line or interoffice facility is out of service, when there is an announcement failure, or when tone receivers are unavailable.

## Automatic Recall (AR)

AR enables the subscriber to direct the switch to place a call to the DN of the last incoming call. When the subscriber dials the AR activation code, the switch retrieves the DN of the last incoming call, and, if the station is idle and calls are allowed to terminate on that line, sets up the call. If the station is busy, the switch monitors the busy/idle status of the calling party's station and of the called party's station for a specified time period (5 to 30 minutes). Scanning for a change in status of the DN to be recalled is performed at the terminating switch if possible, and if not possible, at the originating office through the sending of periodic TCAP requests to the terminating switch. When both stations are idle, the switch applies distinctive ringing to the calling party's line. If the calling party's station is also assigned the calling number delivery (CND) option, the number of the station to be called displays. If the calling party goes off-hook in response to the distinctive ringing, the switch attempts to place the call.

Up to 30 AR requests can be queued for each subscriber. A request is removed from the queue either when the switch places the call to the called party or when the monitoring period for that request expires. The subscriber can cancel all AR requests by dialing a special deactivation code. The subscriber can continue to originate and receive calls without affecting the queued requests.

Because the subscriber usually does not know the identity of the party being re-called or whether the call to be placed is a toll call, the operating company may offer “two-stage” activation of the AR feature. This causes the switch to deliver a digitally-generated voiceback announcement of the DN to the subscriber before attempting to place an AR call. The subscriber then either dials “1" to instruct the switch to proceed with the call or hangs up to cancel the call.

The following conditions apply to AR:

- Although the DMS-10 switch allows up to 30 requests in each station's originating queue, only one request, either ACB or AR, is allowed for a single destination directory number. In addition, only 30 requests are allowed in each station's originating queue for ACB and AR combined.
- Line status checks and queuing are performed only on the called station and not on the directory number hunt (DNH) group to which the station belongs.
- If AR privacy is configured (see Overlay CNFG (FEAT), prompt ARPR) and the calling party has a private number (CNB), the AR subscriber will not be able to recall the number and will be routed to a recorded announcement.

### Automatic Recall (AR) - Block to Private Repetitive

The AR - Block to Private Repetitive feature allows the operating company to prohibit an automatic recall attempt to a DN marked as a private number. This capability is controlled through Overlay CNFG. When a call terminates at a station assigned the Automatic Recall option, call processing checks the call's privacy status. If the DN is private and if the operating company has the AR Block to Private Repetitive blocking bit activated, the AR attempt is blocked and the subscriber is routed to an announcement (VDRA pre-recorded announcement number 72) that informs the subscriber about the privacy status of the calling DN.

In addition, this feature allows the operating company to prohibit multiple AR call attempts to the DN of the last incoming call. This capability is also controlled through Overlay CNFG. Prior to this enhancement, repeated AR calls could be made if the initial call was not completed due to busy condition, no answer, or call completion failure. With the AR block repetitive blocking bit activated, an AR call is considered completed whenever a subscriber is first able to place an AR call to the DN and receives either ring back or busy signal; the call does not need to be answered to be considered completed. If a subscriber attempts to place an AR call to a blocked DN, the subscriber is routed to an announcement (VDRA pre-recorded announcement number 45) that informs the subscriber about the call attempt denial.

The following conditions apply to AR Block to Private Repetitive:

- Only one MADN group member is able to make an AR call attempt to the DN associated with the last call placed to the primary MADN. When the AR block repetitive blocking bit is activated, all MADN members are prohibited from making repetitive AR calls.
- Only one DN on an M5000-Series business set is able to place an AR call to the DN associated with the last call placed to the set. All DNs on the set are prohibited from making repetitive AR calls when the AR block repetitive blocking bit is activated.
- The ability to block repetitive AR call attempts does not prevent the subscriber from making repetitive calls. Other CLASS features, such as CND or ACB, allow a subscriber to call back regardless of AR blocking.
- If a subscriber doesn't answer any of the recall notifications before the recall timer has expired, the AR call attempt is not considered completed.
- AR requests that are blocked due to a private DN restriction or due to repetitive requests are not billed in usage-sensitive billing.

### Automatic Recall for ISDN subscribers

The Automatic Recall feature is available to ISDN subscribers. The feature is activated and deactivated by the subscriber either by dialing AR activation and deactivation codes or by operating a feature activator (single keystroke or other similar action) at an ISDN terminal. There are two types of feature activator: toggle and paired. The toggle feature activator enables one feature activator to be used to activate and deactivate (toggle) AR. The paired feature activator provides two feature activators: one feature activator is used to activate AR while the other feature activator is used to deactivate all requests made by its paired feature activator. ISDN terminals also support a visual feature indicator for AR feature activators, lighting a lamp associated with the service.

**ISDN AR call processing** When an AR call is placed to a DN to be recalled and the DN is not busy, the call is processed immediately. When the call is placed to a DN that is busy, the request is queued and the DN is monitored for a predetermined period of time (5 to 30 minutes). Up to 30 AR requests from each ISDN interface can be queued, regardless of the number of terminals assigned to the interface. When the targeted DN becomes idle, the calling party is notified, even if the party is busy, and the target DN displays on the calling party's terminal. If the calling party goes offhook in response to the notification, the call is placed.

The following conditions apply to ISDN AR:

- The target DN is always displayed during idle notification, whether or not the AR subscriber's terminal is assigned the CND option.
- Dialing an AR deactivation code cancels only the requests made using the activation code; feature activator requests are not affected.

### Office-wide Automatic Recall (OAR)

OAR extends Automatic Recall to subscribers on an office-wide basis. Thus, subscribers do not have to subscribe to the AR feature in order to use it. The operating company can provide one-stage or two-stage OAR to all eligible stations in the office except multiparty lines and coin lines.

Single-party subscribers in the office who do not already have flat-rate billing for Automatic Recall are billed for Automatic Recall on a per-use basis. Flat-rate AR subscribers who use the OAR feature are billed on a flat-rate basis.

When the OAR feature is installed in an office, the feature can still be denied for individual subscribers.

## Calling Name Delivery (CNAM)

The CNAM feature enables a subscriber to view the name, date, and time of an incoming terminating call before answering. After ringing is applied to the subscriber's line, the DMS-10 switch transmits the calling party's name, and the date and time of the call, to the subscriber's station. If the calling party's name is *private* (unavailable for display), or if the name is not available due to an error condition, or if it is associated with an out-of-area number (outside the subscriber's CLASS calling area), a special parameter is sent instead of the calling party's name. Although subscribers' display units vary in the treatment of this parameter, *Private* is normally displayed for a private name and *Out-of-area* is normally displayed for an unavailable or out-of-area name.

Operating companies may restrict display or transmission of calling name information on an office-wide basis. If Originating Calling Name Suppression (prompt ONAS in the SYS prompting sequence of Overlay CNFG) is configured, the calling name display status for all calls originating in the office will be marked *private*. If Terminating Calling Name Suppression (prompt TNAS in the SYS prompting sequence of Overlay CNFG) is configured, the calling name display status for all calls terminating in the office will be marked *private*.

The CNAM feature is available on either a flat-rate or usage-sensitive billing basis. The subscriber cannot deactivate CNAM when it is installed with the flat-rate billing option. If, however, CNAM is installed with the usage-sensitive billing option, the subscriber can toggle the feature on and off by dialing a designated access code.

The following conditions apply to CNAM:

- If Calling Number Delivery (CND) is activated in addition to the CNAM option on the called party's line, the DN of the caller can be displayed along with the caller's name, depending on the capabilities of the called party's display unit.
- Calling name information is delivered to the called party's station during the first long, silent interval (at least 3 seconds in length) of the ringing cycle. If the called party answers the call during the first application of ringing and before the first long, silent interval, normal ring trip will occur, but no data will be transmitted from the terminating switch to the called party's telephone equipment.
- The same access codes are used for activation and deactivation of the usage-sensitive Calling Number Delivery (UCND) and Usage-Sensitive Calling Name Delivery (UNAM) features. Thus, entering a common access code activates/deactivates display of the calling number and the calling name, simultaneously, if both calling information display features are assigned to the station.

- If the terminating switch is unable to transmit calling information to a subscriber's station, the CNAM call terminates as a normal, non-CNAM call.
- CNAM is available only for single-party, residential lines and for EBS subscribers
- CNAM information cannot be delivered if the called party is off-hook.
- CNAM is not provided for Emergency Service Bureau (ESB) service.
- Usage-sensitive CNAM may not be assigned either to automatic or to manual lines because activation/deactivation codes cannot be dialed; CNAM may, however, be assigned to the lines on a flat-rate basis.

### Calling name display control

### SCP calling name data base

This data base contains the subscribers' DNs, names, and the *permanent privacy indicators* associated with the DNs and names. It is the permanent privacy indicator that determines whether the status of a name and number is *presentation restricted* and thus cannot to be delivered, or *presentation allowed* and thus can be delivered, in response to a calling identity feature request.

### Calling Name Delivery Blocking (CNAB)

Calling Name Delivery Blocking enables subscribers to control the display of their names on the called party's display unit, on a per-call basis. When the calling subscriber enters the CNAB activation code, the display status normally associated with the subscriber's name is changed: if the subscriber's name is marked *private* in the SCP calling name data base, the subscriber's name is marked *public* for that call and can be displayed on the called party's display unit.

Calling Name Delivery Blocking is available on both a station basis and an office-wide basis. Office-wide Calling Name Delivery Blocking (ONAB) enables all lines with ANI capabilities in the office, including coin and multiparty lines, to use Calling Name Delivery Blocking.

Calling Name Delivery Blocking is available on either a flat-rate or usage-sensitive billing basis. ONAB is billed only on a usage-sensitive basis.

> ***Note:*** *An incoming call from another office over a SIP trunk does not allow the passing of CNAB information in regards to calling name display. This is due to a limitation in the SIP protocol. When the calling number is public the calling name status is determined by the default status in the calling name database. When the calling number is private the calling name status will always be private as well. It is recommended that the Calling Identity Delivery and Suppression (CIDS) service access codes be used for calls routed over a SIP trunk.*

### Deny Calling Name Blocking (DNAB*)*

When Office-wide Calling Name Delivery Blocking (ONAB) is assigned in an office, the Deny Calling Name Blocking (DNAB) station option may be assigned stations that are to be denied access to the Calling Name Delivery Blocking feature. Attempts to enter the CNAB access code from stations assigned the DNAB station option result in standard error treatment.

DNAB can be assigned to any line with ANI capabilities, including coin and multiparty lines.

### Calling Identity Delivery and Suppression (CIDS)

The CIDS station option can be assigned to any line with ANI capabilities. The CIDS option enables subscribers to set the display status of both calling name and calling number delivery for a single call. When the Calling Name and Number Delivery (CNND) access code is entered followed by a valid directory number, the display status for the calling party's name and number is marked *public* and the calling name and number are delivered to the called party. When the Calling Name and Number Blocking (CNNB) access code is entered followed by a valid directory number, the display status for the calling party name and number is marked *private* and a privacy parameter is delivered to the called party. The CNND and CNNB access codes cannot, however, override office-wide calling identity delivery controls that make names and numbers private (OSUP, TSUP, ONAS, and TNAS prompts in the SYS prompting sequence of Overlay CNFG). CIDS can be billed on a usage-sensitive basis.

CIDS is also available on an office-wide basis (station option OCID). OCID can be assigned to any line with ANI capabilities, including coin and multiparty lines. Assignment of CIDS on both an office-wide basis and on a station basis in an office is allowed.

> ***Note:*** *An incoming call from another office over a SIP trunk does not allow the passing of CNNB/CNND information in regards to calling name display. This is due to a limitation in the SIP protocol. CNNB/CNND apply to calling number status only. When the calling number is public the calling name status is determined by the default status in the calling name database. When the calling number is private the calling name status will always be private as well. It is recommended that the CIDS access codes be used for calls routed over a SIP trunk.*

### Deny Calling Identity Delivery and Suppression (DCID)

When Office-wide Calling Identity Delivery and Suppression (OCID) is configured in an office, the Deny Calling Identity Delivery and Suppression (DCID) station option may be assigned to stations that are to be denied access to the Calling Identity Delivery and Suppression (CIDS) feature. Attempts to enter a CIDS access code from stations assigned the DCID station option result in standard error treatment.

DCID can be assigned to any line with ANI capabilities, including coin and multiparty lines.

### Office-wide name suppression

The Office-wide Originating Calling Name Suppression (prompt ONAS in the SYS prompting sequence of Overlay CNFG) or Terminating Calling Name Suppression (prompt TNAS in the SYS prompting sequence of Overlay CNFG) features may be configured in an office in order to enforce name privacy throughout the office. ONAS marks the name information for all calls originated in the office, including out-going ISUP calls, as *private*. ONAS has no effect on calls originated in the office that route over a SIP trunk group. This is due to a limitation in the SIP protocol. The final name status will be determined by the default status in the calling name database. TNAS marks all terminating calls, including local and incoming ISUP calls, as *private*.

### Calling name display based on calling number display status

A special office-wide parameter may be set in the terminating office to block the delivery of the calling party's name to the called party when the display status of the calling party's number is *private*. To activate this parameter, the terminating office enters *YES* in response to prompt BQPN in the DISP prompting sequence in Overlay CNFG (*Data Modification Manual*, 297-3601-311). When the parameter is activated, SCP name data base queries are blocked and a privacy parameter rather than a name is delivered to the called party if:

- the display status of the calling party's number is *private* and
- the calling party has not activated either Calling Name Delivery Blocking (CNAB) or Calling Identity Delivery and Suppression (CIDS) and
- office-wide name suppression (ONAS or TNAS) is not activated.

If the parameter is not activated (that is, NO is entered in response to prompt BQPN in the DISP prompting sequence in Overlay CNFG [*Data Modification Manual*, 297-3601-311]), the SCP name data base can be queried and the calling party's name can be delivered to the called party depending on the display status of the calling name.

### Interaction of calling name delivery features with other features

**Call Forwarding** Calling identity blocking features (CNAB, CNB, and CIDS) and call forwarding features may be assigned to the same stations. CNAB and CIDS activation codes may be entered before a call forwarding code and remote DN are entered; however, these codes affect only the call which sets up the call forwarding and do not affect the display of the name or number when a call is forwarded.

When a base station that has CNAM active forwards a call to a remote DN, the base station doesn't receive calling party name information for the forwarded call; the calling party name information is forwarded with the call to the remote DN. If the remote DN has calling identity capabilities, it receives calling information based upon the calling information display statuses associated with the original calling party rather than those of the base station. When a call is not forwarded (CFW busy when the line is idle), the calling name information is delivered to the base station.

**Automatic and manual lines** CNAB and CIDS may not be assigned either to automatic or to manual lines because activation/deactivation codes cannot be dialed. DNAB also may not be assigned either to automatic or to manual lines.

**Call Waiting** The delivery of calling name information to an offhook party is not supported. Therefore, a party with Call Waiting receives a call waiting tone but no calling party information.

When rering occurs as a result of the called party disconnecting while a call is waiting, the calling party name information is not sent.

**Calling Identity Delivery and Suppression (CIDS)** CIDS may be used to either deliver or suppress both the number and name of the calling party when CND and/or CNAM are assigned to the terminating station. When the terminating station is an M5000-Series business set and CND is not assigned, use of the CIDS code does not affect the display of number information.

**Calling Number Delivery (CND)** The CND and CNAM options, when assigned to the same station, together determine whether only number information, only name information, or both number and name information is delivered to the called party's station. In addition, delivery of both calling number and name can take longer than delivery only of the calling number because of the time required for obtaining name information through a TCAP query placed to the SCP calling name data base.

A subscriber may not override Office-wide Originating Number Suppression (OSUP) using the Calling Name and Number Delivery (CNND) activation code.

**Calling Number Delivery Blocking (CNB)** CNB may not be used in conjunction with Calling Identity Delivery Suppression (CIDS) because the display status of the calling number (or name) may be changed only once per call; both of these features change the display status of the calling number.

There are no interactions between CNB and CNAM. CNB may be used to toggle the privacy status of the calling party's number when CND is assigned to the terminating station. When the terminating station is an M5000-Series business set and CND is not assigned, use of the CNB code does not affect the display of number information.

**Distinctive ringing patterns** Calling name information is delivered to the called party's station during the first long, silent interval (at least 3 seconds in length) of the ringing cycle. To ensure that all calling party information can be delivered to the called party's display equipment without interruption caused by ringing, it is preferable to assign CNAM to stations that do not also have features requiring distinctive ringing patterns.

**Simplified Message Desk Interface** CNAM may not be assigned to an SMDI line because such a line is not equipped to accept name information. Because the SMDI may receive number information, CIDS can be used to control number delivery to the SMDI.

**Speed Calling** The CNAB or CIDS access code and a DN is a valid short- or long-speed calling list entry. When such an entry is activated, the subscriber is not provided a second recall dial tone but the call proceeds normally.

**Suspended service** All types of suspended service may be assigned to stations that are assigned CNAM, CIDS, CNAB, or DNAB. When complete suspended service is applied, CNAM, CNAB, and CIDS may not be used. When originating service is suspended, UNAM may not be activated or deactivated, and CNAB and CIDS may not be used. When terminating service is suspended, CNAM may not be used.

**Teen Service** When a station assigned CNAM receives a call from a station assigned Teen Service, the name associated with the primary DN is delivered to the CNAM station.

## Canadian Calling Name Delivery

Canadian Calling Name Delivery is a modified version of the DMS-10 Calling Name Delivery feature that can operate in the Canadian telephone network.

Operation of the Canadian Calling Name Delivery feature differs, primarily, in two ways from the operation of the Calling Name Delivery feature in the United States. First, whereas in the United States the terminating switch is responsible for supplying calling name characters for the called party's customer premises equipment, in Canada the originating switch must supply the calling name characters. Second, although both Bellcore (in the United States) and Stentor (in Canada) require that some interoffice call calling name information must be sent in a CCS7 ISUP message from the originating switch, Stentor requires that calling name characters sent in an ISUP message must be contained in a *party information* parameter, while Bellcore requires that the calling name characters must be contained in a *generic name* parameter.

### Interoffice ISUP Canadian CNAM call

The basic operation of interoffice Canadian Calling Name Delivery is illustrated in Figure 2-1. The illustration shows that after the originating party places a call, the originating DMS-10 switch obtains calling name information from the calling name database through a TCAP query. The originating switch then sends an IAM message to the terminating switch. A special *per-ISUP-route* flag (introduced with the Canadian Calling Name feature), manipulated through prompt SPIP in Overlay ROUT (ROUT), determines whether the IAM message will contain the calling name information in a *party information* parameter, the calling name information in a *generic name* parameter, or no calling name information. If the calling name information is to be sent in a *party information* parameter, and the calling name information is blocked or unavailable, the IAM message does not contain any calling name information when sent. If the calling name information is to be sent in a *generic name* parameter, and the calling name information is blocked or unavailable, the generic name parameter indicates that the name is “presentation restricted (P)” or “out-of-area/unavailable (O).” If no calling name information is to be sent, the IAM message sent to the terminating switch does not contain any calling name parameter even if the calling name information is available and presentation of the calling name information is allowed.

Calling name data received in an IAM message is displayed at a terminating party's customer premises equipment only if the terminating party has CNAM activated and if the terminating line is idle. Canadian interoffice calls that utilize a Generic Name parameter send either “O” or “P”, or calling name characters to the customer premises equipment, depending on the information received in the generic name parameter. A Canadian DMS-10 switch that receives a party information parameter displays the name characters in that party information parameter unless the DMS-10 switch blocks calling name delivery. A Canadian DMS-10 switch that receives an IAM message without any calling name information sends an “O” to the terminating customer premises equipment.

### Interoffice SIP Canadian CNAM call

The basic operation of interoffice SIP Canadian Calling Name Delivery is the same as for ISUP Canadian Calling Name Delivery as shown in Figure 2-1. When a call is routed over a SIP trunk a SIP INVITE message is sent rather than an ISUP IAM. Differences arise in the information available in an INVITE message versus the ISUP IAM.

The SIP protocol does not provide for the transmission of an independent name and number privacy status. When the calling number privacy status is *private*, the calling name status is also *private* regardless of the calling name status in the database. When the calling number status is *public* and the calling name status is *private*, no name characters are included in the INVITE message resulting in an *unknown* name status. When the calling number and calling name status are *public*, the calling name characters are included in the INVITE message and are displayed to the terminating customer premises equipment.

### Intraoffice Canadian CNAM call

The basic operation of intraoffice Canadian Calling Name Delivery is illustrated in Figure 2-2. The illustration shows that after the originating party places a call, the originating switch sends a TCAP query to the calling name database. After the switch receives the calling name information, the originating party's calling name is delivered to the terminating customer premises equipment if the terminating party has CNAM activated, the terminating party is idle, and calling name delivery is not blocked. No ISUP signaling is required in this case, since the call terminates in the originating office.

**Figure 2-1: Interoffice Canadian Calling Name Delivery Call Processing**

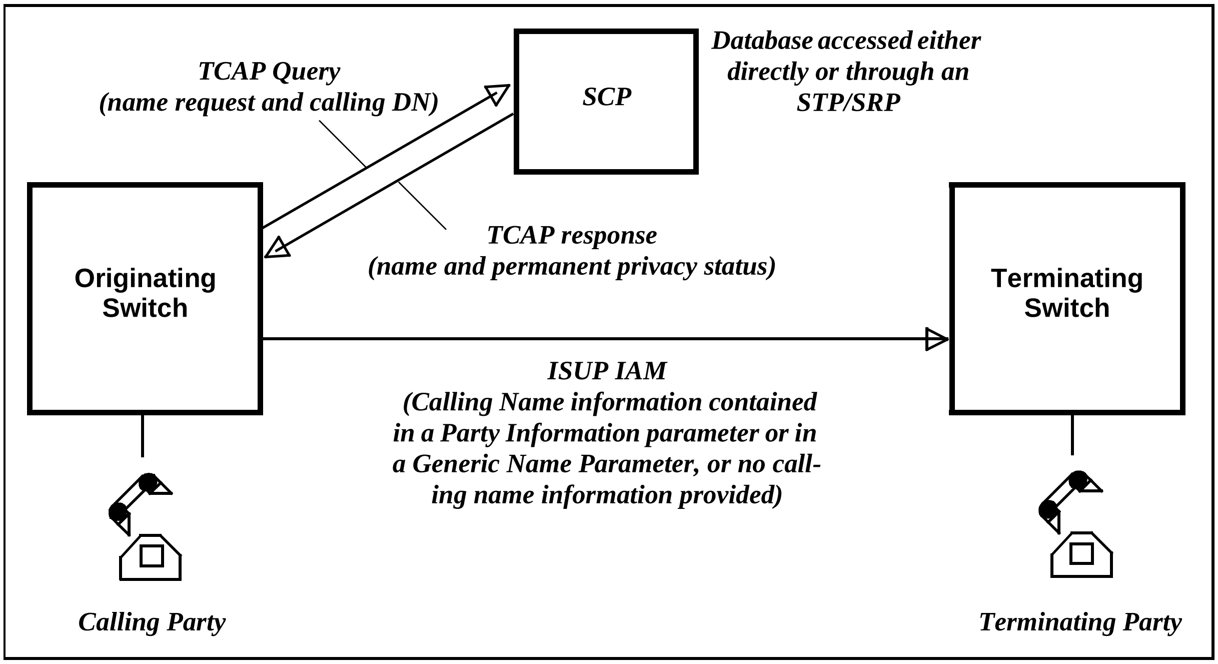
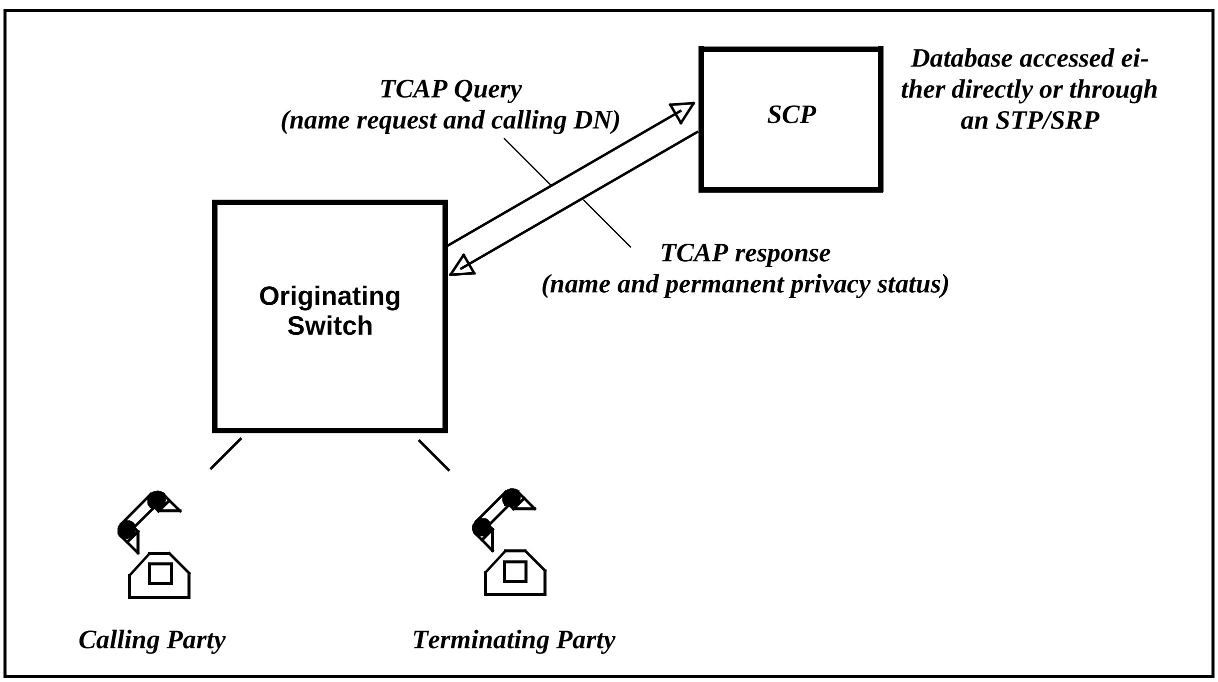



**Figure 2-2: Intraoffice Canadian Calling Name Delivery Call Processing**



## Calling Number Delivery (CND)

The CND feature enables the subscriber to view the DN of an incoming terminating call before answering. After ringing is applied to the subscriber's line, the DMS-10 switch transmits the calling party's 10-digit DN and the date and time of the call to the subscriber's station. If the calling party's DN is “private” (unavailable for display), the DMS-10 switch will send a “private” indicator instead of the DN. If the calling party's DN is either an out-of-area number (outside the subscriber's CLASS calling area) or a partial number (less than 10 digits long) the DMS-10 switch will send an “out-of-area” indicator instead of the DN.

If CND is installed with the subscription billing option, the subscriber cannot deactivate the CND option. If, however, CND is installed with the usage-sensitive billing option, the subscriber can toggle the feature on and off by dialing an access code.

The following conditions apply to CND:

- (U)CND cannot be assigned to multi-party or manual lines. In addition, CND service cannot be denied to groups of lines based on class-of-service.
- For offices that are not part of a CCS7 network, “O”, rather than a DN, is delivered for inter-office calls.
- PINs, or four-digit Personal Identification Numbers, cannot be delivered in the place of a DN.
- The calling party's DN cannot be delivered to a (U)CND station while the subscriber at that station is off-hook.
- A calling party's DN can only be transmitted to a forwarded station that has the CND option.
- When a station ringer test is performed on a (U)CND line, a test pattern, rather than the DN, is delivered to the subscriber's station.
- The calling party's DN is never transmitted to an SMDI mailbox.

### Dialable Number Delivery (DND)

Dialable Number Delivery (DND) is an enhancement of the Calling Number Delivery (CND) feature. DND causes only the digits that must be dialed in order return the call to display rather than all ten digits normally displayed by CND. The DND display always takes precedence over the CND display.

DND is available to subscribers on either a flat-rate billing basis or on a usage-sensitive billing basis. DND can be assigned only to subscriber lines that are also assigned the CND/UCND station options.

## Calling Number Delivery Blocking (CNB)

CNB enables subscribers to control the display of their DNs on the called party's station, on a per-call basis, by dialing the CNB activation code. If an individual station is assigned the SUPR (calling number delivery suppression) station option, dialing the CNB activation code causes the DN to be displayed. If, on the other hand, the station is not assigned the SUPR station option, dialing the CNB activation code causes “P” (private) to be displayed rather than the DN. CNB may be assigned either on a station or an office basis.

Although the caller's DN can be blocked from display by CNB and by station or office-wide CND suppression, the DN is still transmitted to the terminating CLASS end office. Therefore, the called party can still use the Automatic Recall, Customer Originated Trace, and CLASS screening features.

The following conditions apply to CNB:

- If a station is in an office with office-wide calling number delivery suppression (prompt OSUP = YES in overlay CNFG (SYS)), all forms of calling number delivery blocking (OCNB, CNB) are inoperable.
- A CNB subscriber does not also have to have the CND option in order to block DN display on outgoing calls.
- (U)CNB cannot be assigned to multi-party or manual lines. In addition, CNB service cannot be denied to groups of lines based on class-of-service.
- When a CNB subscriber activates CND suppression, or, if the subscriber's line is also assigned the CND suppression station option (SUPR), the calling party's DN will not be delivered to SMDI lines.

## Customer Originated Trace (COT)

COT enables a subscriber to initiate a trace on the last incoming call. The trace information, including the traced DN, the date and time of the trace, the location and DN of the called party's station, and the date and time of the call to be traced is either printed at the operating company's office or transmitted to a law enforcement agency. COT may be installed either as a station option or as an office-wide option. If installed as an office-wide option, COT can be billed either by subscription or according to usage.

When the subscriber activates COT, the switch first verifies that the DN it has recorded for tracing is complete and not ambiguous, that is, the call didn't originate at a PBX interface or multiparty line, before proceeding. If the DN is not valid, a partial trace is generated and the switch instructs the subscriber to contact the local operating company for additional assistance.

The operating company may offer “two-stage” activation to guard against accidental activation of a trace. After dialing the COT activation code, the subscriber must either dial “1" to activate the trace, or hang up to cancel the trace.

## Screening List Editing (SLE)

SLE enables the subscriber to configure and operate the four selective call screening services, Selective Call Forwarding (SCF), Selective Call Rejection (SCR), Selective Call Acceptance (SCA), and Selective Distinctive Ringing/Call Waiting (SDR). SCF, SCR, SCA, and SDR can all be billed either by subscription or according to usage.

SLE also enables the Simultaneous Ringing (SRNG) subscriber to configure and operate the SRNG feature. SRNG is not a CLASS feature but is included here because it uses the SLE functionality in the same way as the selective call screening services. SRNG is offered on a subscription basis only.

After the subscriber enters the access code for one of the services, the switch provides the subscriber with the status of the service (active/inactive), the number of DNs (public and private) in the screening list associated with the service, and instructions pertaining to the service being accessed. The subscriber may then change the status of a service, query or change one of the screening lists, or ask for instructions pertaining to the selected service.

### Selective Call Forwarding (SCF)

This feature enables the subscriber to designate incoming call DNs that are to be forwarded. The subscriber also designates a remote DN to which the incoming calls from designated DNs are to be forwarded when SCF is active. The subscriber can either enter a DN directly into the list of SCF DNs or direct the switch to add the DN from the last incoming call to the list. Thirty-two DNs can be stored in the SCF list.

### Selective Call Rejection (SCR)

This feature enables the subscriber to specify incoming call DNs that are to be denied termination. Rejected calls are given a prerecorded call rejection announcement and are terminated. The subscriber can either enter a DN directly into the list of SCR DNs or direct the switch to add the DN from the last incoming call to the list. Thirty-two DNs can be stored in the SCR list.

### Selective Call Acceptance (SCA)

SCA enables the subscriber to have incoming calls screened against a specified list of DNs. Only calls from stations with DNs found in the list are allowed to terminate on the subscriber's line. The rejected calls are given a prerecorded call rejection announcement and are terminated. The subscriber can either enter a DN directly into the list of SCA DNs or direct the switch to add the DN from the last incoming call to the list. Thirty-two DNs can be stored in the SCA list.

### Selective Distinctive Ringing/Call Waiting (SDR)

This feature enables the subscriber to designate up to 32 directory numbers from which incoming calls are to be identified by distinctive ringing or, if the subscriber also has the Call Waiting (CWT) feature, by a distinctive call-waiting tone. The subscriber can either enter a DN directly into the list of SDR DNs or direct the switch to add the DN from the last incoming call to the list.

### Simultaneous Ringing (SRNG)

This feature is listed here even though it is not a CLASS feature and as such not all of the SLE interactions apply to the SRNG feature. The Simultaneous Ringing feature and its interactions are described in detail later in this section. The Simultaneous Ringing feature does use the Screen List Editing function to enable the SRNG subscriber to designate a list of DNs in the same manner as SCA, SCF, SCR, and SDR. When the SRNG subscriber receives an incoming call, the numbers on the SRNG list will also be alerted. The subscriber can either enter a DN directly into the list of SRNG DNs or direct the switch to add the DN from the last incoming call to the list. Four DNs can be stored in the SRNG list.

> ***Note:*** *The DN of the last incoming call may be added to the SRNG list of a physical SRNG DN only. The last calling DN cannot be added to the SRNG list for a VDN.*

### Screening List Editing features for ISDN subscribers

The selective call services associated with the Screening List Editing feature are available to ISDN subscribers. The Simultaneous Ringing feature is not available to ISDN subscribers. The features can be assigned only to DNs with the VI (voice and voiceband information, speech and 3.1 -kHz audio bearer capabilities) call type.

**Processing screening list commands and digits** SLE features available for ISDN subscribers will not support the processing of multiple digits or a called party element received within an ISDN message. In either event, the subscriber is given an announcement indicating that too many digits were entered and is returned to either the main list editing level or the previous list editing level, depending on the activity being performed when the digits were received. If multiple digits are received when the subscriber is attempting to activate SCF through the use of a feature activator, the subscriber is re-prompted to enter the RDN.

**Adding entries to the screening list** The verification of the number entered by an ISDN subscriber includes a check to ensure that the number has the VI call type. If the number has a call type other than VI, the subscriber receives an error announcement and must begin again.

**Accessing a screening list by multiple TEIs of a DN/CT** Only one access by a TEI of a DN/CT to obtain a service status report, or to create or modify the status of a screening list, is permitted at one time.

**SLE feature activators and indicators** Each of the SLE features can be activated (call-associated or non-call associated) using feature activators (single keystroke or other similar action) at an ISDN terminal. The same feature activator is used to activate and deactivate (toggle) its associated selective call service. ISDN terminals also support visual feature indicators for each of the selective call services, lighting a lamp associated with the active selective call service and extinguishing the lamp when the service is deactivated.

### Voice Back Blocking

This feature enables the service provider to mark all calls that terminate in a DMS-10 office as “private” in the called stations' incoming memories, thus preventing either delivery or voicing back of the caller's number.

The following conditions apply to the interaction of the Voice Back Blocking feature with other CLASS features:

- All directory numbers added to screening lists (SLE features) by using the ‘01' command (‘add last incoming call') will be marked ‘private' in the lists.
- All calls terminating to a CND subscriber's station will be marked ‘private' and will not be displayed on the subscriber's display unit.
- AR subscribers with two-stage activation will not have the targeted number voiced back prior to AR activation; the announcement will indicate, instead, that AR is activated for a private number.
- All calls terminating to CLASS subscribers in an office configured with Voice Back Blocking will be marked ‘private' in the called stations' incoming memory regardless of activation of CNB by the originator of the call.
- All calls terminating to CLASS subscribers in an office configured with office-wide originating call suppression will be marked ‘private' in the called stations' incoming memories.

## Simultaneous Ringing

The Simultaneous Ringing (SRNG) service, available with Generic 504.10 and later 500-Series generics, allows a user-defined group of up to five Directory Numbers (DNs) to be alerted at the same time. The SimRing group comprises a single Pilot DN (PDN) and up to four Non-Pilot Member DNs (NPMDNs). The first alerted DN that answers the call is connected to the calling party, while the calls to the other alerted member DNs are released.

The Pilot DN must be a subscriber served by a DMS-10 switch. The NPMDNs can be either a DN in the same DMS-10 switch with the PDN or can be a Wireline or Wireless DN served by another switch.

SimRing groups are treated as multi-user groups on call termination. A call to a SimRing group PDN only receives busy treatment when all reachable member DNs are busy. Otherwise, idle member DNs receive simultaneous notification and the calling party receives audible power ringing. Actual member DN notifications (power ringing, Call Wait tones) may not be simultaneous. Member DN notification depends on the amount of time required to establish a connection with each member DN. A noticeable delay can occur, for example, for cellular NPMDNs.

The SimRing Virtual Directory Number (VDN) feature allows the SimRing option to be added to a Pilot DN that does not have a physical line card appearance on the switch. In this way the SimRing functionality can be offered to subscribers not served directly by the DMS-10.

The SimRing service allows the subscriber to activate or deactivate the service on the PDN as well as edit the list of NPMDNs via the Screen List Editing (SLE) functionality. The SLE functionality is accessed by dialing a service access code which connects the subscriber to a voice prompted, menu-driven user interface to guide subscribers through the list of available options. NPMDNs can be any valid dialable DN in the North American Public Switched Telephone Network (NA PSTN) except for intraswitch MADN. Interoffice trunking between the DMS-10 and other NPMDNs may use inband, ISUP or ISDN PRI trunks.

SimRing can also be set up so that the subscriber can access the SLE functionality remotely from any location through an Access DN (ACDN). The use of Personal Identification Number (PIN) codes provide the end user with the necessary security protection. When a subscriber wishes to access the SimRing service remotely, he/she dials the ACDN access number. Once the call is connected, the subscriber will dial the PDN followed by the PIN code. Following an additional tone prompt, the subscriber is then able to dial the SimRing service access code to access the SLE functionality.

The simultaneous alert occurs when the Pilot DN receives a call and the feature is active. When a single party NPMDN is on the same DMS-10 as the PDN, it will be alerted using a distinctive ring pattern for SimRing calls. When the call is to a Virtual DN busy treatment is provided.

**Voice Mail Interactions** The following conditions apply to Voice Mail/Call Forwarding and SimRing.

It is desirable that busy and unanswered calls to the SimRing group should be forwarded to and answered by one Voice Mail System (VMS), rather than that of the first member DN to answer the call. The DMS-10 will identify one member in the group to be the SimRing voice mailbox as follows:

- When the PDN has voice mail, busy calls to the SimRing group PDN will be forwarded to and answered by the PDN's Voice Mail System. The subscriber will then only have to check their PDN voice mailbox for messages.

- When the PDN does not have voice mail or service is provided via a VDN, the DMS-10 will expect that the first member in the list has voice mail and that busy calls to the SimRing group PDN should be forwarded to and answered by this member's VMS. However, calls are still offered to the other non-pilot member DNs, and so it is left to the subscriber to ensure that the NPMDN which has voice mail is in the first position in the list. The subscriber will then only have to check this voice mailbox for messages, unless the other NPMDNs also have voice mail.

To help ensure that voice mail messages to the SimRing group are offered only to the PDN or first NPMDN, when any non voice mail NPMDN call is routed out of the office on an ISUP trunk, the maximum number of redirected calls parameter in the IAM will be set to prevent any forwarding from occurring in the terminating office. However, for non-ISUP trunks, this cannot be enforced and so any inter-office NPMDN which has any forwarding active to a voice-mail system will result in this system answering the SimRing call (answer indication will be sent on the incoming trunk).

If any inter-office NPMDNs do have a voice mail option, the first busy NPMDN which forwards to the voice mail system may cause an answer of the SimRing call. This is true in particular, for calls to member DNs which are routed over an inband trunk or when ISUP-to-inband interworking is encountered. For member DNs local to the switch that is hosting the PDN which are beyond the first position in the list, the DMS-10 will not forward on the busy condition.

**Call Forward Don't Answer Timers**
For calls which are forwarded to a voice mail system on no-answer, the forwarding timer represents the number of rings after which a call is forwarded. The SimRing subscriber should manage these timers so that the timer value associated with the PDN (or first NPMDN when there is no forwarding from the PDN) should be less than the timer values associated with the other NPMDNs.

**Mobile NPMDNs**
When an NPMDN is a mobile terminal and that terminal is either ‘out of area' or turned off, the call will be automatically redirected to the mobile service providers VMS or to the provider's announcement system. In such a case it is likely that the mobile service provider's VMS or announcement system will answer the SimRing call before a user is able to answer the call at another location. The SimRing subscriber should manage the SimRing group list to remove the mobile NPMDN from the list while the mobile terminal is out of area or turned off.

**Busy SimRing group member DNs**
A call to a SimRing group PDN with the voice mail service who is currently engaged in another call will be redirected to the PDN's voice mailbox. When the PDN has voice mail, the Call Forward Busy (CFB) feature will have precedence over the SimRing service, and none of the NPMDNs will be alerted.

When an NPMDN with the CFB option is currently engaged in another call (i.e., busy condition), the PDN voice mail option is examined before the NPMDN call is forwarded. If the PDN resides on the same switch and has the voice mail option then the call to the NPMDN is released. If the PDN does not have the voice mail option or if the PDN resides on another switch, the call to the NPMDN is attempted and the Call Forward Busy feature is invoked. In such a case, it is possible that the NPMDN service providers VMS will answer the SimRing call before a user is able to answer the call at another location.

### Distinctive Alerting

The PDN will be alerted using its regular alerting patterns; for example, normal alerting, distinctive ringing, teen service, long distance alerting, etc. The exact type of alerting depends upon feature subscription.

When a single-party NPMDN is on the same DMS-10 as the PDN and the SimRing group PDN receives a call, the NPMDN will be alerted using a distinctive alerting pattern. When the NPMDN subscribes to a distinctive alerting feature (distinctive ringing, teen service, long distance alerting, etc.) that alerting will take precedence over the SimRing distinctive alerting. Multi-party NPMDN lines shall continue to be alerted using their normal alerting patterns.

When a NPMDN is on a different switch from the PDN, the terminating switch does not know whether the call originated from a SimRing group PDN or was placed directly to the NPMDN. This can lead to the SimRing call being answered by someone other than the intended SimRing user. A solution may be to assign a Secondary DN (e.g., teen service) to the NPMDN and use the Secondary DN as the SimRing NPMDN to distinguish (by different alerting treatment) between SimRing calls and direct calls to the NPMDN.

## Telemarketer Call Screening

The Generic 504.10 Telemarketer Call Screening (TELE) feature attempts to eliminate telemarketer calls to a subscriber by screening suspected Telemarketer calls before ringing the subscriber. The TCS service intercepts calls that are suspected to be Telemarketer calls. The service plays an announcement stating that the party they have dialed does not accept calls from telemarketers, and that the party wishes for his/her name to be added to the telemarketer's ‘‘Do Not Call" list. Other callers are asked to dial ‘1' to ring through.

## Virtual Directory Number

The Generic 504.10 Virtual Directory Number (VDN) feature allows a Directory Number (DN) to be assigned on a DMS-10 switch. Although it has no physical line card appearance, a VDN will have a virtual line appearance. It is administered as a single-party line in the station (STN) prompting sequence in overlay DN as well as in other overlay prompting sequences

In a DMS-10 switch, a virtual DN is assigned as a seven-digit or ten-digit DN in the North American Numbering Plan (NANP). A ten-digit DN is used when the Duplicate NXX feature is configured in the switch (prompt DNXX = YES in Overlay CNFG (SYS)) and the thousands group (NXX X) specified has more than one associated HNPA. Up to 2048 virtual directory numbers can be assigned.

In Generic 504.10, a Virtual DN may be used to offer the following services:

### Alarm Dispatch

The Alarm Dispatch (ALDP) feature allows for a service provider to administer a call out list for alarm reporting in the office. In the event of an alarm, the DMS-10 switch automatically calls the first directory number (DN) in the list. The technician at that DN may or may not acknowledge the alarm. If the alarm is not acknowledged, the DMS-10 calls the next DN in the list, and so on, until the alarm is acknowledged. A VDN is used to place the above calls.

### Simultaneous Ringing

The (SimRing) feature allows the SimRing station option (SRNG) to be added to a SimRing group Pilot DN (PDN). A virtual PDN (for example, a VDN with the SRNG station option) is used to offer SimRing service to subscribers in the local calling area who are hosted off of a switch that does not provide SimRing service. The subscriber has remote access to the user interface by dialing an Access DN (ACDN).

> ***Note:*** *The DN of the last incoming call may be added to the SRNG list of a physical SRNG DN only. The last calling DN cannot be added to the SRNG list for a VDN.*

### Remote Call Forwarding

A VDN assigned as a Remote Call Forwarding Appearance (RCFA) provides remote call forwarding. However, a VDN has an advantage over the existing RCFA service offered because the subscriber has remote access to the service and can activate, deactivate, and change the forward-to number.

### Advanced Intelligent Network

A DN may be assigned as an Advanced Intelligent Network (AIN) VDN. In the DMS-10, AIN VDNs are currently defined using the AIN Termination Attempt (TA) trigger assigned to an RCFA.

## Web Based Feature Control

The Generic 505.10 Web Based Feature Control (WEBF) allows subscribers to maintain their CLASS, Simultaneous Ringing, and Speed Call lists via a third party interface. List update are made through a web page that the service provider maintains. The service provider enables some or all of the features assigned to the subscriber on the DMS-10 for update via the Web page.

The DMS-10 will interface to the subscriber's web interface through a intermediate SCP/Adjunct. The DMS-10 interfaces to the SCP with Common Channel Signalling (CCS) links. The Service Node (SN) will send the DMS-10 instructions to update the subscriber lists using TCAP Packages as defined in Telcordia GR-1299 AINGR: Switch - Service Control Point (SCP)/Adjunct Interface.

Specifically, a subscriber would be able to make the following changes to their CLASS and SimRing lists:

- Activate the feature
- Deactivate the feature
- Add a number to the list
- Add the last calling party to the list
- Add a speed calling entry to the list
- Delete a number from the list
- Delete all private entries from the list
- Delete all entries from the list
- Query the list entries
- Query the list size
- Query feature activation status

Additionally for Selective Call Forwarding the user can:

- Set the redirection number
- Set the redirection number to a speed call entry
- Clear the redirection number
- Update the redirection number

For Long Speed Calling, Short Speed Calling, the subscriber can:

- Set an index
- Query their speed call list

The WEBF feature will have not impact on the billing. The DMS-10 will not track messages sent to the DMS-10 for billing purposes.

## Multiple Appearance Directory Number

The Multiple Appearance Directory Number (MADN) feature enables a DN to be assigned to a residential subscriber consisting of up to eight members forming a MADN group. The MADN group may be configured both with 500/2500-type sets and with Voice over IP (VoIP) terminals. Each member of a MADN group may be assigned a different set of station options.

MADN groups are configured in a Single-Call Arrangement (SCA). In this configuration, only one MADN member may be active (either originating or terminating) on the MADN at a given time.

For a procedure used to set up the MADN feature, see SOP 0135 in NTP 297-3601-311, *Data Modification Manual*.

### MADN call processing

To originate a call from a MADN group, the entire MADN group must be call processing idle and the originating number must be in service. If the MADN is busy, the subscriber placing the call receives five seconds of reorder tone. If the MADN is not busy, the subscriber receives dial tone, indicating that the call can be placed.

When a MADN is called, ringing is started for all in-service members of the group. Any member of the MADN group may answer the call. When the call is answered, ringing stops on all of the other MADN group members.

### MADN Call Hold on 500/2500-type sets

A MADN group member may place a call that has been answered on hold by performing a switch-hook flash, dialing the “MADN hold” access code, and then going on-hook.

Any 500/2500 type member of the MADN group can then access the MADN by going off-hook.

### MADN Call Hold on Voice over IP (VoIP) terminals

For Voice over IP (VoIP) terminals, placing a call that has been answered on hold will depend on the VoIP terminal. For example, when a MADN group member is using a 2500-type set connected to a terminal adapter, the user would follow the steps described above. When the VoIP terminal has multiple line or DN keys, A MADN group member would press the DN key, dial the “MADN hold” access code, and then going on-hook.

Any VoIP member of the MADN group can then access the MADN by going off-hook (dial tone is provided by the VoIP terminal) and dialing the “MADN hold” access code to retrieve the held call.

### Conditions applying to MADN VoIP SIP terminals

The following conditions apply to the MADN feature:

- Because a proceed to dial indication such as dial tone is provided by the VoIP Session Initiation Protocol (SIP) device when the user goes off-hook or presses a line appearance key, the MADN group member must dial a “MADN hold” access code in order to retrieve a held call. The MADN hold and cancel hold access codes must be defined in translations. For details about the translations, refer to overlay TRNS in the NTP entitled *Data Modification Manual* (297-3601-311).
- When a SIP gateway line is assigned as the primary MADN group member and the Call Forward on Internet Down (CFID) feature is assigned and currently active, then calls terminating to the MADN DN will be forwarded when the primary MADN group member’s SIP user agent is not registered; no MADN members will be alerted, including any wired lines. For more information see "Call Forward Internet Down (CFID)" on page 8-4.
- When a SIP gateway line is assigned as the primary MADN group member, then any features active on the user's gateway, such as call forwarding or do not disturb, take precedence over the alerting of the secondary MADN group members. Only when the gateway responds that it is alerting will the DMS-10 proceed to alert the other MADN group members.
- When a SIP gateway line is assigned as a secondary MADN group member, then any features active on the user's gateway, such as call forwarding or do not disturb, will cause that member to be passed by.

### Conditions applying to Calling Line Id on MADN terminals

Calling Line Id (CND - Calling Number Delivery and CNAM - Calling Name Delivery) may be assigned to the primary MADN group member with 500/2500-type sets and SIP terminals, and to secondary MADN group members with SIP terminals.

# Section 3: Business services

## Introduction

This section presents general descriptions of the services of interest to business subscribers served by a DMS-10 central office. Most of these services are defined in the DMS-10 software as station options. For additional information, see the NTP entitled *Data Modification Manual* (297-3601-311).

## Integrated Business Services (IBS)

> ***Note:*** *The IBS feature package is marketed as "Multiline Variety Package" (MVP). The term IBS and the mnemonics for IBS features are still used throughout the NTPs and in the DMS-10 software.*

IBS allows a business to integrate up to six lines into a single customer group. The DMS-10 switch can support up to 255 IBS groups (numbered 001 through 255). The DNs in a given IBS group may have different office codes as long as calls between these office codes do not require 1+7-digit dialing. An IBS group must use such single-party lines as 1FR (flat-rate), 1MR (residential message-rate), or 1MB (business message-rate) and can select a combination of the IBS features for each of its member stations.

When a station is added to an IBS group, it is assigned an IBS group member number. The member numbers range from 2 through 7; 1 is not used. IBS group members are assigned member numbers according to the order in which they are added to the IBS group: the first station added to the IBS group is assigned member number 2, the second station is assigned member number 3, and so on. When a station is deleted from the group, its member number is not reassigned to any existing members of the group; rather, its number is assigned to next station added to the group.

In generics 602.20 and later, SIP Gateway lines may be added to IBS groups.

For a procedure used to add the IBS option to a station, see SOP 0025 in NTP 297-3601-311, *Data Modification Manual.* To determine if the DMS-10 switch is configured with IBS, see overlay CNFG (SYS) in NTP 297-3601-311, *Data Modification Manual.*

## IBS access codes

Some of the features in the IBS package require the use of feature access codes. These codes are configured through overlay TRNS (PRFX) prompting sequence. Translations can be configured for any set of codes the operating company requires, although standard access codes are recommended by Bellcore for operating companies not requiring customized access codes.

## Intercom (INT)

The INT feature allows an IBS user to call other stations within the same IBS group by dialing an octothorp (#) followed by the INT code of the called station. The call will be put through immediately, and any subsequent digits will be ignored.

When a station is added to an IBS group, it is added to the group's *intercom* table and is assigned an INT code. A station's INT code corresponds to its IBS group member number, one of the digits 2 through 7, assigned to the station when it is added to an IBS group. Because each station added to the IBS group is assigned the next available IBS group member number, operating company personnel must query each IBS group member (using overlay DN (STN) in NTP 297-3601-311, *Data Modification Manual*), in order to determine the INT code assigned to the individual member stations. Although all IBS groups have an intercom table, only those stations with the INT feature can use it for INT calls.

The following conditions apply to the INT feature:

- Except for SIP subscribers, a station assigned the INT feature must also be assigned the Digitone (DGT) option. For SIP lines, DGT is not compatible.
- The INT feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the INT feature.
- Some SIP devices may not support Intercom dialing, or a digit map change on the device may be required to allow the Intercom dialing pattern.

## Call Hold (CHD)

The CHD feature allows a member of an IBS group to place an established call on hold by flashing the switchhook and dialing the CHD feature code. This frees the IBS user's line to originate another call, use Call Pickup, retrieve a waiting call, or return to a previously held call.

The CHD feature may be initiated by a properly classmarked subscriber who is either a party in an established two-way call or a non-controlling member of an established three-way conference call. The IBS user performs a switchhook flash, which gives the IBS user confirmation tone followed by dial tone. While receiving dial tone, the IBS user may perform one of the following actions:

- No action-Dial tone is removed after 10 seconds, and the Digitone receiver is released. Dial tone may then be recalled by another switchhook flash. Dial tone is sent for 10 seconds each time it is recalled. The IBS user may return to the second party (or conference-call parties) at any time by performing a disconnect or by dialing the CHD feature code while receiving dial tone.
- Retrieve a waiting call-The user may retrieve a waiting call by performing a switchhook flash and dialing the CHD feature code.
- Initiate another call-If the IBS user initiates a call to a third party, he may return to the second party (or conference-call parties) by performing a disconnect or by performing a switchhook flash and dialing the CHD feature code.
- If the IBS user performs a switchhook flash and then dials the CHD feature code during an established two-way call with a third party, the third party is placed on permanent hold, and the IBS user is reconnected to the second party. The IBS user may then alternate between the second and third parties through subsequent switchhook flashes followed by the CHD feature code.
- Initiate a Call Pickup-The IBS user may initiate a Call Pickup by dialing the Call Pickup feature code.
- Disconnect-If the IBS user performs a disconnect while the second party (or conference-call parties) are on permanent hold, the IBS user will receive a rering signal. Upon answering, the IBS user is connected to the second party (or conference-call parties).

The following conditions apply to the CHD feature:

- Except for SIP subscribers, a station assigned the CHD feature must also be assigned the Digitone (DGT) option. For SIP lines, DGT is not compatible.
- The CHD feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the CHD feature.
- The Local Coin Overtime / Custom Calling Interface feature enables parties to utilize the CHD feature when called from a coin phone. For additional information about the LCO feature, refer to Section 5 of this NTP.

## Call Pickup (CPU)

The CPU feature enables an IBS user (assigned the CPU station option) to answer a call to an unattended station (assigned the CPU station option) within the IBS user's group. The CPU feature may be invoked while the IBS station is in either an idle condition or a talking connection.

To invoke CPU from an idle condition, the IBS user goes off-hook, receives regular dial tone, and then dials the CPU feature code. This connects the IBS user with the unanswered call. If there is more than one unanswered call, the IBS user is connected to the one that has been ringing the longest.

To invoke CPU from a talking connection, the user must be assigned the Call Hold feature. The user places the current call on Call Hold, receives dial tone, dials the CPU feature code, and is connected as above.

After being connected with the unanswered call, the IBS user may perform one of the following actions:

- Disconnect-If a previous call is on Call Hold, the IBS station is rung and upon answer is connected to that call.
- Switchhook flash-The IBS user may return to the previous call or alternate between calls by following the procedures or retrieving a call placed on Call Hold.
- Transfer-If the IBS user has been assigned the User Transfer feature, the user may transfer the picked-up call, whether or not another call is on permanent hold. The call may be transferred by flashing the switchhook, receiving special dial tone, dialing the destination code, and going on-hook (either during ringing or after answer). If a previous call has been placed on permanent hold, the IBS station is rung again and, upon answer, is reconnected to the previous call.

The following conditions apply to the CPU feature:

- Except for SIP subscribers, a station assigned the CPU feature must also be assigned the Digitone (DGT) option. For SIP lines, DGT is not compatible.
- The CPU feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the CPU feature.

## Directed Call Pickup Without Barge-in (DCPU)

The DCPU feature allows a member of an IBS group to answer a call to another station within the IBS group. The operation of DCPU is similar to Call Pickup operation except that the call pickup is directed to a specific station.

The Directed Call Pickup features are controlled by a feature bit. With this feature bit enabled, the IBS group member has the option of receiving all Directed Call Pickup features, which include DCPU, DCBI, DPUA, DCBX, and DCPX.

DCPU may be invoked while the IBS station is in either an idle condition or a talking connection. To invoke DCPU from an idle condition, the user goes off-hook, receives regular dial tone, and dials the DCPU feature access code. After receiving special dial tone, the user dials the IBS station's intercom (INT) code. This connects the user with the unanswered call, unless the call has been abandoned or already answered. If the call has been abandoned or already answered or the called station has been assigned the Directed Call Pickup Exempt option, the user will receive generic busy treatment (usually busy tone). Regular dial, special dial, and busy tones are defined in the NTP entitled *System Performance Specifications* (297-3601-180).

To invoke DCPU from a two-way talking connection the user must be assigned the Call Hold feature. The user places the current call on hold, receives dial tone, and invokes DCPU as though the line is idle.

After being connected with the incoming call, the user may disconnect from the call, perform a switchhook flash, or transfer the call. For more information on these actions, refer to the description of the Call Pickup feature.

DCPU may not be used like an intercom call. If DCPU is invoked and the station to which the call is directed is idle, the user will receive generic busy treatment (usually busy tone).

The following conditions apply to DCPU:

- When the called station is assigned to a hunt group, DCPU will override the hunting sequence.
- Convenience Dialing (CVD) may be used with DCPU by dialing the DCPU feature code followed by the station's two-digit CVD code.
- DCPU cannot be used to pick up a call that is being forwarded to another station. However, DCPU can be performed on the station to which a call has been forwarded.
- When the called station has been assigned the Don't Answer Transfer (DAT) option and DCPU is not performed within the DAT timeout period, the call will be transferred and the DCPU user will receive generic busy treatment.
- Both the station assigned the DCPU feature and the station to which DCPU is directed must be assigned the INT option.
- DCPU cannot be invoked when the user is already engaged in a three-way call.
- If the DCPU target station is a SIP Gateway line, and the SIP line is active on another call or calls as a source party (originator), a terminating call to the SIP line can be picked up.  However, if the target SIP line is active on any other calls as a destination party, an additional terminating call cannot be picked up.

- The DCPU feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the feature.

### Directed call pickup with barge-in (DCBI)

DCBI allows a member of an IBS group to answer a call to another station within the IBS group and to barge-in if the call is already established. The operation of DCBI is the same as DCPU operation except that barge-in allows the DCBI user to be connected in a three-way call with the two parties that are already in a talking state. The DCBI feature access code is the same as the DCPU feature access code.

The Directed Call Pickup features are controlled by a feature bit. With this feature bit enabled, the IBS group member has the option of receiving all Directed Call Pickup features, which include DCPU, DCBI, DPUA, DCBX, and DCPX.

Before the DCBI user barges in, a barge-in tone may be given to the two subscribers who are already connected. DCBI tone provision is configured for the DMS-10 switch in overlay CNFG (CP prompting sequence). If the barge-in is not allowed (see the conditions listed below), the DCBI user will receive generic busy treatment (usually busy tone). Regular dial, special dial, busy, and barge-in tones are defined in the NTP entitled *System Performance Specifications* (297-3601-180).

The conditions listed for DCPU apply to DCBI and the following conditions also apply to DCBI:

- Barge-in is allowed only on a call that is established at the called station. For example, if the call has been picked up by another group member, barge-in is not allowed.
- Barge-in is allowed only on a call that is in a normal two-way talking state. For example, if the subscriber at the called station has activated three-way call or call hold, barge-in will not be allowed.

  ***Note:*** *If a SIP Gateway line initiates a three-way call, this restriction does not apply as the SIP device is responsible for the 3WC in this case, and from the perspective of the DMS-10, the Gateway line is in a normal two-way talking state.*

- Barge-in is not allowed when the called station has been assigned the Directed Call Pickup Barge-In Exempt option.
- DCBI has precedence over Call Waiting (CWT). If a three-way call has been created by a barge-in, incoming calls to both the called station and the station where DCBI was invoked will receive generic busy treatment even if those stations have the CWT option assigned.
- DCBI has precedence over Busy Transfer (BTF, BTFA, or BTFI). If the called station has a Busy Transfer option assigned and the incoming call has been answered, barge-in will occur.

- In a DMS-10 Classic Network configuration, a Conference pack (NT4T03) must be configured in the DMS-10 switch to provide the three-way call after barge-in.
- The DCBI feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the feature.
- The DCBI feature permits emergency override and monitoring by the EBS attendant.

## Directed Call Pickup Barge-in Exempt (DCBX)

DCBX is a terminating station option that prevents any attempt by another member of the IBS group to barge in on an answered call. Calls to the station where DCBX has been assigned may be picked up from a station that is a member of the IBS group and has been assigned the CPU or DCPU option or the DCBI option (if the call has not been answered). The DCBX feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the feature.

The Directed Call Pickup features are controlled by a feature bit. With this feature bit enabled, the IBS group member has the option of receiving all Directed Call Pickup features, which include DCPU, DCBI, DPUA, DCBX, and DCPX.

## Directed Call Pickup Exempt (DCPX)

DCPX is a terminating station option that prevents any attempt by another member of the IBS group to pick up a call to the station by using DCPU or DCBI. Calls to the station where DCPX has been assigned may be picked up from a station that is a member of the IBS group and has been assigned the CPU option. The DCPX feature is assigned on an IBS-station basis in Overlay DN. This overlay allows the assignment, deletion, and query of the feature.

The Directed Call Pickup features are controlled by a feature bit. With this feature bit enabled, the IBS group member has the option of receiving all Directed Call Pickup features, which include DCPU, DCBI, DPUA, DCBX, and DCPX.

## Directed Call Pickup from Any Station (DPUA)

DPUA is a terminating station option that allows calls to the station to be picked up from any other station in the IBS group. The operation of DPUA is the same as that for DCPU except that the DPUA feature access code must be dialed instead of the DCPU feature access code. The DPUA feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the feature.

The Directed Call Pickup features are controlled by a feature bit. With this feature bit enabled, the IBS group member has the option of receiving all Directed Call Pickup features, which include DCPU, DCBI, DPUA, DCBX, and DCPX.

## Busy Transfer (BTF)

The BTF feature allows calls from outside the IBS group routed to a busy station in the IBS group to be rerouted automatically to another station (the target DN) within the group. If the target DN is also busy, the call is transferred to the BTF target DN designated for that station. This process is repeated to a maximum of five transfers. If all stations in the IBS group are busy, a busy condition is signaled to the incoming call.

The following conditions apply to the BTF feature:

- A station cannot be assigned both the Call Waiting and Busy Transfer features.
- Since SIP lines use local client-based call waiting, incoming calls to a SIP line with the BTF feature assigned will continue to be presented to the SIP line until the client device returns a '486 Busy Here' indication. For example, if the SIP client supports two call appearances and client-based call waiting is enabled, BTF features will not operate until the client has one call active and one call waiting. If the client supports more than two call appearances through additional line keys, BTF features will not operate until all call appearances are in use and the SIP client returns a busy indication to the next incoming call.
- A station assigned the BTF feature cannot also be assigned to a hunt group.
- The BTF feature is assigned on an IBS-station basis in overlay DN. The BTF target DN *(nnn nnnn)* is assigned at that time and is modifiable only through DMO. The station user is not allowed to modify the BTF target DN directly. Overlay DN allows the assignment, deletion, and query of the BTF feature.
- The BTF target DN must be a member of the same IBS group as the transferring station.
- An IBS group member station may not be deleted if it is a BTF target DN for any other group member.

## Busy Transfer All (BTFA)

The Busy Transfer All feature (BTFA) is an enhancement to BTF. It is identical to BTF except that BTFA allows any calls that terminate in an IBS group to be rerouted automatically to another station (the target DN) within that group when a busy condition is encountered.

BTFA is enabled by a feature bit. Once the feature bit is enabled, the station option may be assigned.

The conditions that apply to BTF apply to BTFA. Also, a line may be involved in the calling sequence only once.

## Busy Transfer Intragroup (BTFI)

The Busy Transfer Intragroup (BTFI) feature is an enhancement to BTF. It is identical to BTF except that BTFI allows only calls that originate and terminate in the same IBS group to be rerouted automatically to another station (the target DN) within that group when a busy condition is encountered.

BTFI is enabled by a feature bit. Once the feature bit is enabled, the station option may be assigned.

The conditions that apply to BTF and BTFA also apply to BTFI.

## User Transfer (UTF)

The UTF feature enables an IBS user to transfer an established call to another line.

For 500/2500 type terminals, an IBS user can transfer an established call to two subscribers by performing a switch-hook flash, dialing the third party to which the call is to be transferred, and performing a disconnect at any time following completion of dialing. For SIP lines, the call transfer is performed using instructions for the particular SIP client device.

The following conditions apply to the UTF feature:

- An IBS station assigned the UTF feature is allowed to form a three-way conference call even if it is not classmarked for three-way conference.
- A party outside the controlling party's IBS group cannot be transferred to another party outside the IBS group.
- A user cannot initiate a user transfer or a three-way call while a call is waiting unless the user is classmarked for the Call Hold feature as well as the UTF feature.
- A user cannot receive Call Waiting after initiating a three-way call.
- A user cannot transfer a party on Call Hold or Call Waiting until that party has been retrieved.
- After system initialization, a two-party connection will be rebuilt between the IBS user and a party not on hold. Any parties on hold will be dropped.
- The UTF feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the UTF feature.
- The Local Coin Overtime / Custom Calling Interface feature enables parties to utilize the UTF feature when called from a coin phone. For additional information about the LCO feature, refer to Section 5 of this NTP.

- If the DMS-10 switch is not responsible for providing ringback (that is, early cut-through for user-provided audible ring is active on the PRI (prompt UPAR = YES in Overlay PRI (PRI)), then the transfer cannot be performed before the telephone at the PRI answers.

## Call Transfer Outside (CTO)

The CTO feature enables all IBS subscribers with user transfer capability to transfer an established call to two subscribers outside of the IBS group.

The following conditions apply to the CTO feature:

- All IBS groups with UTF have CTO capability (see prompt CTOI in Overlay CNFG (CCS) in NTP 297-3601-311, *Data Modification Manual*)
- All SIP Gateway lines in IBS groups have CTO capability, using the SIP client-based call transfer capability. The DMS-10 UTF option is not compatible with SIP lines.

## Don't Answer Transfer (DAT)

The DAT feature is an IBS line option that allows a terminating call to an idle IBS line to be automatically transferred to another predesignated line within the IBS group. The transfer destination is specified by service order when the DAT option is assigned to an IBS line and cannot be changed or deactivated by the subscriber.

When a call is transferred, the call originator may hear an interruption in ringing. However, the originator will continue to receive audible ringing until the transfer destination answers or a ringing timeout occurs. At the transfer destination, a DAT call and any other incoming calls are indistinguishable.

The call is transferred if it is not answered at the called line within a preselected number of ringing cycles. The number of ringing cycles, which can range from 2 to 10 (in increments of 1), is specified in Overlay CNFG (CP) (see NTP 297-3601-311, *Data Modification Manual*) and applies to all IBS groups assigned to the switch.

If any of the following conditions exist, calls terminating on an IBS line with DAT are not transferred and continue to ring the called station until calls are abandoned or answered:

- The maximum number of call legs (six) has been reached
- The transfer destination is in a busy condition and
  - does not have CWT and is not a SIP Gateway line, or
  - has CWT and a call is already waiting, or
  - is a SIP line and returns a Busy Indication.

- The transfer destination has the INWATS feature and the call is an intraoffice call
- The transfer destination has the Denied Terminating or Suspended Service features
- The transfer destination is originating the call, or the call has just been transferred from the transfer destination
- The transfer destination has activated Call Forwarding to a DN that is not in the IBS group have the BTF feature.

If the transfer destination (DAT target) has call forwarding active, the call is transferred regardless of the state of the call forwarding destination. The originator receives the appropriate treatment, including busy tone, if the call forwarding destination is in call-processing-busy state.

The following condition pertains to the DAT feature: when the User programmable Call Forward Busy Don't Answer (CFBD) option is activated on a line that is assigned both CFBD and DAT, DAT is disabled.

## Three-Way Calling (3WC)

The 3WC feature allows an IBS user to form a three-way conference call with two other parties located either within or outside the IBS group. To do so, the IBS user performs a switchhook flash during a normal talking connection, receives special dial tone, and dials a third party. Following completion of dialing, the user performs a switchhook flash to create a three-way conference. A subsequent switchhook flash drops the added party and re-establishes the original connection. If the IBS user performs a disconnect during a three-way call, the entire call is dropped or, if the IBS user is assigned the UTF feature, the original call is transferred to the added party, if one of the two remaining parties is a member of the same IBS group as the user.

The following conditions apply to the 3WC feature:

- If, while in a normal talking state with the third party, the user attempts to form a three-way conference and no conference circuit is available, the third party will be placed on temporary hold and the user will be reconnected to the second party. The user may then alternate between the second and third parties through subsequent switchhook flashes until a conference circuit becomes available.
- If a user attempts to form a three-way conference before a normal talking state is reached with the third party and no conference circuit is available, the third party is dropped and the user is reconnected to the second party.
- A user cannot form a three-way conference while a call is waiting unless the user is classmarked for the Call Hold feature as well as the 3WC feature.

- A user cannot receive Call Waiting indication after initiating a three-way call.
- A user cannot form a three-way conference involving a party on Call Hold or Call Waiting.
- After system initialization, three-way conference calls will not be rebuilt.
- The 3WC feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the 3WC feature.
- The DMS-10 3WC option is not assigned to SIP Gateway lines. The SIP device provides conference functionality locally. Follow the instructions for the SIP device in order to perform a conference call from a SIP line.
- If a three-way call is in progress and equipment involved in the call, other than the NT4T03 Conference pack or NT8T04 Network Interface pack, is removed from service, the call is disconnected.
- The Local Coin Overtime / Custom Calling Interface feature enables parties to utilize the 3WC feature when called from a coin phone. For additional information about the LCO feature, refer to Section 5 of this NTP.

### Three-Way Calling enhancement

The operation of the Three-Way Calling feature has been enhanced in order to comply with Bellcore requirements found in document TR-TSY-000577, *Three-Way Calling*. Prior to this enhancement, a party placed on consultation hold would be disconnected if the party initiating the three-way call disconnected before a three-way call was established. With this enhancement, the party on consultation hold will not be disconnected and the switch will ring back the party initiating the three-way call so that the call can be re-established; ring-back continues either until the party answers or until the party on consultation hold disconnects.

The following conditions apply to the 3WC feature enhancement:

- Processing of Local Coin Overtime events continues while the switch is ringing back the party initiating the three-way call.

## Call Forwarding (CFW)

The IBS CFW feature is identical to the Custom Calling Services feature.

## Call Forwarding DMO Activation/Deactivation

The IBS Call Forwarding DMO Activation/Deactivation feature is identical to the Custom Calling Services feature.

## Call Forwarding Limitation (CFL)

The IBS CFL feature is identical to the Custom Calling Services feature.

## User Programmable Call Forward Busy Don't Answer

The IBS User Programmable Call Forward Busy feature is identical to the Custom Calling Services feature.

## Usage Sensitive User Programmable Call Forward Busy Don't Answer

The IBS Usage Sensitive User Programmable Call Forward Busy Don't Answer feature is identical to the Custom Calling Services feature.

## Call Waiting (CWT)

The CWT feature alerts a user who is busy on an existing two-way call that another call is waiting. The CWT feature is initiated when a call from inside or outside the IBS group arrives at a properly classmarked IBS station that is busy on a two-way call. The calling party is connected to normal audible-ringback tone, and the IBS user receives the CWT tone.

After receiving the CWT tone, the IBS user may ignore the waiting call or retrieve the waiting call in one of the following ways:

- Disconnect-The existing call is dropped, and the IBS user receives an appropriate ring signal from the waiting party. The waiting party receives normal ringback. When the IBS user answers, the connection to the waiting party is completed.
- Switchhook flash-If the called IBS user is not classmarked for the Call Hold feature (CHD), he may retrieve the waiting call by performing a switchhook flash. This places the previously connected party on consultation hold, and the IBS user is connected to the waiting party. The IBS user may then alternate between the two parties through subsequent switchhook flashes.
- Call Hold feature code-If the called IBS user is classmarked for the CHD feature, the waiting call may be retrieved by performing a switchhook flash, receiving special dial tone, and dialing the CHD access code. This places the previously connected party on call hold, and the IBS user may then alternate between the two parties by performing subsequent switchhook flashes, receiving special dial tone, and dialing the CHD access code.

The following conditions apply to the CWT feature:

- The DMS-10 CWT option is not compatible with SIP Gateway lines. The SIP device provides call-waiting functionality locally. Follow the instructions for the SIP device in order to use the call-waiting function.
- A controlling party may perform a user transfer or form a three-way conference call while a call is waiting if the user has been assigned the CHD or UTF option.

- The CWT feature can be initiated when the called IBS user is either the called or calling party of the original call.
- A station cannot be assigned both the CWT and Busy Transfer (BTF, BTFA, or BTFI) features.
- The CWT feature may be used only when the IBS station is in a normal talking state.
- Only one call can wait or be on hold for a busy IBS user. Subsequent callers will receive busy treatment.
- The CWT feature is deactivated for the controlling party of a three-way conference call.
- CWT treatment of calls forwarded is based on the class of service of the terminating line.
- Normal ringing time out processing will be used for a waiting call.
- After system initialization, a two-party connection will be rebuilt between the IBS user and the party not on hold. The party on hold will be dropped.
- A user may not perform a CPU while having a Call Waiting.
- The CWT feature is assigned on an IBS-station basis in Overlay DN. This overlay allows the assignment, deletion, and query of the CWT feature.
- The Local Coin Overtime / Custom Calling Interface feature enables parties to utilize the CWT feature when called from a coin phone. For additional information about the LCO feature, refer to Section 5 of this NTP.

### Call Waiting Chaining (CWC)

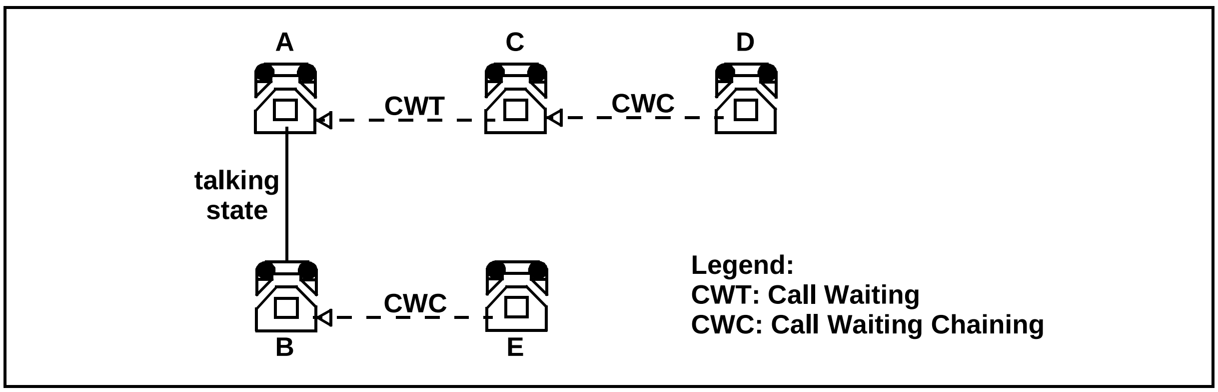
Call Waiting Chaining, introduced in Generic 409.11 (making DMS-10 Call Waiting compliant with Bellcore requirements in TR-TSY-000571), makes Call Waiting treatment available for incoming calls placed to parties whose calls either are being held, are waiting, or are in talking state in a Call Waiting call. As illustrated in Figure 3-1, if a call is established between parties A and B, Call Waiting enables party C's call to party A to be placed in the call waiting state. If, subsequently, party D calls party C, the Call Waiting Chaining (CWC) feature enables party D's call to wait on party C. If, at this same time, party E places a call to party B, Call Waiting Chaining enables party E's call to wait on party B. Up to seven non-controlling Call Waiting call parties (parties whose calls are waiting) can receive Call Waiting Chaining calls.

The following conditions pertain to the Call Waiting Chaining feature:

- When a Call Waiting call is chained, two controlling parties (parties that receive a call waiting signal) can place the connection between each other on hold simultaneously.

- A controlling party in a Call Waiting Chaining call can be placed on hold simultaneously by two parties with which the controlling party is connected.
- Call Waiting Chaining cannot be used by a held party in a three-way call or an IBS/EBS call hold call.
- When Call Waiting Chaining is in effect, a hook-flash can only be used for Call Waiting and will not initiate a three-way call, user transfer, or IBS/EBS call hold.
- During system initialization or system reload (SYSLOAD), only connections between subscribers in stable two-way conversations are maintained.

**Figure 3-1: Call Waiting Chaining**




## Cancel Call Waiting (CCWT)

The CCWT feature is an enhancement to CWT that allows a subscriber with the CWT option to inhibit the call waiting tone for the duration of a single call. CCWT can be made available to the subscriber either as part of IBS CWT or as an additional feature for those who purchase IBS CWT. Refer to the description under “Custom Calling Services” for information on its operation.

### Calling Identity on Call Waiting (CWID)

The CWID feature causes the identity of a calling party in a waiting call to be displayed on the called party's station, unless the identity of the calling party is private or is unavailable for display.

CWID is available to Calling Number Delivery (CND), Calling Name Delivery (CNAM). CWID subscribers are automatically subscribed also to Call Waiting (CWT); however, if the subscribers end subscription to the CWID feature, their subscription to CWT is not affected.

**CWID Interaction with MBS Call Waiting** The caller's ID displays at the time that the Call Waiting lamp is turned on.

**CWID Interaction with ISDN** CWID is incompatible with ISDN terminals.

**Imposed CWT Treatment** If an EBS subscriber assigned a calling line ID option (CND, CNAM) receives a call from a station assigned either Call Waiting Originating (CWTO) or Dial Call Waiting (DCWT), the station receiving the call is automatically assigned Call Waiting (CWT) but not CWID.

**CWID Interaction with Anonymous Call Rejection (ACR)** A subscriber assigned both the CWID and ACR features is not alerted for calls for which presentation is restricted.

**CWID Interaction with Automatic Recall (AR)** Since a calling station must be idle in order to receive AR ring back, no alerting tones indicating an AR attempt are given to a calling party with CWID that is involved in a stable call.

If an AR request applies to a busy station that has CWID assigned, the calling station is given audible ringing and the CWID station is given the CWID treatment for the call; if the call cannot be waited, the calling party is given a busy signal and normal AR processing continues.

**CWID Interaction with E911 Calls** During a call to a 911 attendant, CWID is disabled.

**CWID Interaction with Call Forwarding** Call Forward Busy (CFB) is applied to calls that cannot be waited. Call Forward Busy, Don't Answer (CFD) is applied to calls that receive CWID treatment but are still unanswered after expiration of the CFD time interval.

**CWID Interaction with Call Transfer** Busy Transfer (BTF) is applied to calls that cannot be waited. Don't Answer Transfer (DAT) is applied to calls that receive CWID treatment but are still unanswered after expiration of the DAT time interval.

**CWID Interaction with Cancel Call Waiting (CCWT)** A subscriber can override CWID for a particular call by activating the CCWT feature before placing the call. A subscriber with CWID, CCW, and Three-way Calling (3WC) can activate CCWT during a two-way call before receiving any CWID alerting; if the subscriber flashes the switch-hook during the call, the subscriber receives dial tone and can then dial the CCWT access code to become re-established with the original call.

**CWID Interaction with Selective Distinctive Ringing/Call Waiting (SDR)** A call placed to a station assigned both CWID and SDR from a DN that is not on the station's DR screening list receives standard CWID treatment. If the calling subscriber's DN is on the SDR screening list, the call also receives CWID treatment, but a SDR alert tone is then substituted for the subscriber alert signal.

**CWID Interaction with customer premises equipment (CPE)** Subscribers with the CWID option will hear an additional tone and subsequent momentary disconnect of the speech path while the customer premises equipment transacts the new, incoming calling data for display. The subscriber alert tone will remain unchanged, providing two warning tones 6 to 10 seconds apart.

## Distinctive Ringing (DSR)

The DSR feature applies different ringing patterns to allow members of an IBS group to distinguish between the following two call types:

- Terminating intragroup calls
- Terminating calls from outside the IBS group

Idle IBS lines receiving intragroup call are given a ringing pattern of 2s ring-4s silence. Idle IBS lines receiving call from outside their IBS are given a ringing pattern of .5s ring-1s silence-.5s ring-4s silence. Additionally, busy IBS lines equipped with both the CWT and DSR features receive a different CWT tone for each of these two call types. LCM and RCU lines are given a ringing pattern of .5s ring-.5s silence-.5s ring-4.5s silence.

> ***Note:*** *DSR can be used only with single-frequency ringing. If coded ringing is configured, RNG, TIP, R1, or T1 are the only ring codes that may be assigned. If MF ringing is configured, RNG or TIP are the only ring codes that may be assigned. If SIMP ringing is configured, RNG, TIP, R1, R2, T1, or T2 are the only ring codes that may be assigned.*

The DSR feature is assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the DSR feature.

## Ring Again (RAG)

The IBS RAG feature is identical to the Custom Calling Services feature.

The following conditions apply to the Ring Again feature

- The originator and target must be served by the same switch.
- The originating station must have the RAG station option and the target station must not have the RAG denied station option.
- The originating station can only have one RAG request active at any time.
- The target station can not exceed 16 RAG requests, which are served in the order in which they are received.
- RAG is assigned on an IBS-station basis in overlay DN.
- RAG cannot be invoked against a station that has performed a BTF, BTFA, or BTFI.

- RAG re-ring is not modified when the receiving station has DSR.

### Ring Again Denied (RAGD)

The IBS RAGD feature is identical to the Custom Calling Services feature.

### Convenience Dialing (CVD) and Convenience Dialing Controller (CVDC)

These two interrelated features, CVD and CVDC, enable an IBS group to use and maintain a *convenience dialing list*, which associates each of up to 30 frequently-called numbers with a two-digit CVD code (20 through 49).

The CVD feature allows an IBS user to call one of these frequently-called numbers by dialing the CVD access code followed by the number's CVD code. The frequently-called number is then automatically dialed. The number can consist of up to 15 digits; if Equal Access Abbreviated Dial 3 is configured, the number can consist of up to 20 digits. The number can consist of up to 30 digits.

The CVDC feature, which can be assigned to one or more stations in an IBS group, allows the CVD List to be established and maintained. A user at a designated CVDC station may enter or change a number in the list by performing the following actions:

**1)** Go off-hook and dial the CVD update code.

**2)** After receiving special dial tone, dial the CVD code to be entered or changed.

**3)** Dial the frequently-called number to be associated with the CVD code entered in Step 2.

**4)** Complete the entry by dialing an octothorp (#) or by waiting for an interdigital timeout to occur (see Note). If there has been no dialing error, confirmation tone is given after the entry is complete, and the list is then updated. The user is then given 5 seconds to go on-hook before disconnect timing is invoked.

***Note:*** *The interdigital timing for CVD list updates is as follows:*

- After collecting the CVD code, a long interdigital timeout
- Until seven CVD entry digits are received,a long interdigital timeout
- After seven CVD entry digits are received, a short interdigital timeout

The following treatments are given to dialing errors that may occur during CVD list updates:

- If the user dials an invalid CVD code, an asterisk (*), or more than the number of digits allowed (15 or 20), the call is generically routed to dialing error, and the list is not updated.
- If the user goes on-hook before receiving the confirmation tone, the list is not updated.

The following conditions apply to the CVD and CVDC features:

- Except for SIP subscribers, a station assigned either the CVD or CVDC feature must also be assigned the Digitone (DGT) option. For SIP lines, DGT is not compatible.
- The CVD and CVDC features are assigned on an IBS-station basis in overlay DN. This overlay allows the assignment, deletion, and query of the CVD and CVDC features.

## Enhanced Business Services (EBS)

***Note:*** *The EBS feature package is marketed as "Integrated Business Services" (IBS). The term EBS and the mnemonics for EBS features are still used throughout the NTPs and in the DMS-10 software. Note, however, that the EBS package is different from the IBS feature package (marketed as "Multiline Variety Package") and the two feature packages are not line compatible.*

The EBS feature package offers custom calling type features to multiline business subscribers. EBS allows the business customer to integrate separate access lines into a single communication group without special premises equipment.

A business subscribing to EBS is designated as an EBS customer group. A DMS-10 switch equipped with EBS can support up to 512 groups and up to 4,000 members per group. All lines terminating in the EBS group must be served by a single DMS-10 office or its associated remotes. An EBS group must use such single-party lines as 1FR (flat-rate), 1MR (residential message-rate), or 1MB (business message-rate). Each line can be a member of only one EBS group. A combination of the EBS features may be selected for each member of an EBS group.

EBS basic services are described in the sections entitled *Station-to-Station Calling* and *Direct Outward Dialing*, and descriptions of the optional features follow those paragraphs.

The optional line features can be assigned to lines within an EBS customer group on a per-feature and per-line basis (through overlay DN). The Call Waiting Enhancements features are treated as one feature in the DMS-10 software. An operating company can offer any subset of the optional features as a standard package. The DMS-10 switch also can provide traffic and feature usage measurements for each EBS group.

In generics 602.20 and later, SIP Gateway lines may be added to EBS groups.

## EBS group dialing plan

A unique dialing plan can be assigned to each EBS customer group; however, all lines in an EBS group must share the same dialing plan. An EBS prefix translator, which allows Station-to-Station (STS) calling, Direct Outward Dialing (DOD), and access to optional features, is assigned to each group, and any number of EBS groups can be assigned to the same prefix translator. Different dialing plans can be created for each EBS group by defining unique translators through overlay TRNS, EBSP prompting sequence.

STS calls are not subject to DDD billing used in the local area. No special billing record is required on DOD calls. All DOD-chargeable calls (toll, message-rate, or local call detail recording) are recorded per existing AMA formats on a DN basis. If the Special Billing station option is used, all chargeable DOD calls made by an EBS group will be billed to a single DN within the group.

Direct Inward Dialing (DID) allows stations outside the EBS group to call a specific station within the group by dialing the seven-digit DN assigned to the EBS station. This procedure is subject to EBS station restrictions and all POTS line call-terminating treatments.

## Station-to-station calling (STS)

The STS feature is provided for all EBS customer groups in conjunction with the EBS dialing plan. The feature allows an EBS member to complete a call to other stations within the same EBS group by dialing the last one to five digits of the station's DN. The number of digits to be dialed by members of an EBS group is selected by the customer and normally depends on the number of stations in the group. The STS feature can be used with Call Forwarding, Three-Way Calling, Call Transfer, and Call Hold. Some SIP client devices may not support one-digit STS dialing; two- to five-digit STS dialing patterns are recommended for any EBS group including SIP lines.

To place an STS call, a member of an EBS group goes off-hook, receives regular dial tone, and dials the last one to four digits of the DN of the called station. Upon receipt of the number of digits required for a STS call, the DMS-10 switch will terminate the call as if the seven-digit DN of the called station was dialed. A 4-s timeout will be applied only if there is a conflict in digit translation.

STS calls to unassigned numbers are given vacant code DN generic treatment or the vacant code DN treatment defined for the EBS group. STS calls to temporarily out-of-service numbers are given suspended line treatment. Partial dial timing is initiated after each digit is received unless the DMS-10 switch has determined that STS dialing is complete.

The following parameters apply to the STS feature:

- STS calling between different EBS groups, from EBS lines to POTS lines, and from POTS lines to EBS lines is not allowed.

- STS calling does not apply to Automatic Line, Manual Line, or Denied Originating EBS lines.
- The DMS-10 switch cannot restrict selected EBS group stations from STS calling.

## Direct Outward Dialing (DOD)

The DOD feature is provided for all EBS customer groups in conjunction with the EBS dialing plan. The feature allows stations within a EBS group to place calls to DNs outside the EBS group.

To access the POTS dialing plan, a member of an EBS group goes off-hook, receives regular dial tone, and dials a DOD access code. The EBS group member receives a second dial tone and then dials the external number, which is designated by the POTS dialing pattern.

The following parameters apply to the DOD feature:

- During dialing, the user must pause after the DOD access code until a second dial tone is received.
- A second dial tone does not guarantee that the EBS station will obtain an outgoing trunk.
- DOD can be used with Call Forwarding, User Transfer, Three-Way Calling, and Speed Calling and Group Speed Calling (the DOD call digits must be stored in the speed calling list).
- DOD does not apply to Automatic Line, Manual Line, and Denied Originating EBS lines.

## Call Pickup Group (CPUG)

The CPUG feature allows an EBS station user to answer a call to an unattended station in the same call pickup group. The CPUG feature is similar to the IBS Call Pickup (CPU) feature; however, CPUG introduces call pickup groups, which are subsets of an EBS customer group. ISDN Call Pickup is only functional for lines where the office equipment interface access is configured for speech capability or 3.1kHz audio.

Each EBS customer group can contain up to 50 call pickup groups. An EBS member can belong to only one call pickup group and can use CPUG only within that group. The CPUG feature may be invoked while the EBS station is in either an idle or talking condition.

To invoke CPU from an idle condition, the EBS call pickup user goes off-hook, receives regular dial tone, and then dials the CPU feature code. This connects the EBS call pickup user with the unanswered call. If there is more than one unanswered call, the EBS call pickup user is connected to the one that has been ringing the longest. An ISDN EBS member assigned CPU can answer a ringing phone in the pickup group by originating a voice call and using a CPU feature activator, or by dialing the CPU feature access code. An ISDN EBS member assigned CPU, however, cannot answer a ringing phone in the pickup group if the call is delivered through Additional Call Offering (ACO).

To invoke CPU from a talking connection, the user must be assigned the Call Hold feature (see “Call Hold”). The user places the current call on Call Hold, receives dial tone, dials the CPU feature code, and is connected as above.

After being connected with the unanswered call, the EBS call pickup user may perform one of the following actions:

- Disconnect-If a previous call is on Call Hold, the EBS station is rung and upon answer is connected to that call.
- Switchhook flash-The EBS call pickup user may return to the previous call or alternate between calls by following the procedures for retrieving a call placed on Call Hold (see “Call Hold”).
- Transfer-If the EBS call pickup user has been assigned the User Transfer feature, the user may transfer the picked-up call, whether or not another call is on permanent hold. The call may be transferred by flashing the switchhook, receiving special dial tone, dialing the destination code, and going on-hook (either during ringing or after answer). If a previous call has been placed on permanent hold, the EBS station is rerung and, upon answer, is reconnected to the previous call.

CPUG is an optional station feature that is assigned, deleted, and queried in overlay DN. A call pickup group can be queried in overlay ODQ.

## Directed Call Pickup Without Barge-In (DCPU)

The EBS DCPU feature is identical to IBS DCPU except for the parameters listed below. Refer to “Directed Call Pickup Without Barge-In” under “Integrated Business Services.”

The following parameters apply to EBS DCPU:

- The EBS station's Station-to-Station (STS) code must be dialed instead of the IBS Intercom code to invoke DCPU access.
- DCPU may not be used like an STS call. If DCPU is invoked and the station to which the call is directed is idle, the user will receive generic busy treatment (usually busy tone).

- Group Speed Calling (GSC) may be used with DCPU by dialing the DCPU feature code followed by the station's two-digit GSC index number. Also, the DCPU feature code and a station's STS code may be programmed as a GSC number if neither the feature code nor the station's STS code contain an asterisk (*) or octothorp (#).
- Any virtual facilities associated with a call will remain with the call when it is picked up. If additional virtual facilities become associated with the call, these will be integrated into one call appearance. In the unlikely event that more than six virtual facilities are used, the call will be dropped.
- Except for SIP subscribers, a station assigned the DCPU feature must also be assigned the Digitone (DGT) option. For SIP lines, DGT is not compatible.
- An ISDN EBS member assigned DCPU can answer a call terminating at a specific directory number by originating a voice call and dialing the DCPU feature access code. After hearing a special dial tone, the subscriber is able to enter the specific extension number.

## Directed Call Pickup With Barge-In (DCBI)

The EBS DCBI feature is identical to IBS DCBI except for the parameters listed below. Refer to “Directed Call Pickup With Barge-In” under “Integrated Business Services.”

The following parameters apply to EBS DCBI:

- DCBI tone provision is configured for the EBS group in overlay HUNT (EBS prompting sequence).
- DCBI has precedence over Call Waiting (CWT, CWTI, or CWIG). If a three-way call has been created by a barge-in, incoming calls to both the called station and the station where DCBI was invoked will receive generic busy treatment even if those stations have a Call Waiting option assigned.
- An ISDN EBS member assigned DCBI can initiate DCBI by originating a voice call and dialing the DCBI feature access code. After hearing a special dial tone, the subscriber is able to enter a DCBI subscriber's extension number.

## Directed Call Pickup Barge-In Exempt (DCBX)

The EBS DCBX feature is identical to IBS DCBX. Refer to “Directed Call Pickup Barge-In Exempt” under “Integrated Business Services.” For ISDN subscribers, DCBX is assigned to a terminating DNCT.

## Directed Call Pickup Exempt (DCPX)

The EBS DCPX feature is identical to IBS DCPX. Refer to “Directed Call Pickup Exempt” under “Integrated Business Services.” For ISDN subscribers, DCPX is assigned to a terminating DNCT.

## Directed Call Pickup From Any Station (DPUA)

The EBS DPUA feature is identical to IBS DPUA. Refer to “Directed Call Pickup From Any Station” under “Integrated Business Services.” Any ISDN EBS member can answer a call terminating at an ISDN terminal assigned DPUA by originating a voice call and dialing the DPUA feature access code. After hearing a special dial tone, the EBS member subscriber is able to enter the DPUA subscriber's extension number.

## Don't Answer Transfer (DAT)

The EBS DAT feature is identical to IBS DAT except that the IBS DAT number of rings is assigned through overlay CNFG while the EBS DAT number of rings is assigned through overlay HUNT. IBS DAT is described as part of the “Integrated Business Services (IBS)” features. The DAT feature has been expanded to include Voice and Circuit Mode Data calls offered to ISDN EBS members.

## Speed Calling (SSC and LSC)

The Speed Calling feature allows an EBS member to maintain and use a speed calling list, which associates frequently called numbers with a one- or two-digit index number. The EBS feature package includes two types of speed calling: Short-List Speed Calling (SSC) and Long-List Speed Calling (LSC).

The SSC list consists of up to eight frequently called numbers. An EBS member creates or changes a SSC list by going off-hook, receiving regular dial tone, dialing the SSC update list access code , receiving special dial tone, and dialing a SSC index number followed by the digits to be associated with that index number. A confirmation tone indicates that the list has been updated. The SSC list is used by dialing the perform SSC call access code followed by the single digit associated with the desired DN. The user may dial an octothorp (#) or wait for an interdigital timeout to occur.

The LSC list consists of up to 30 frequently called numbers. The procedures for creating and using a LSC list are identical to those for creating and using a SSC list, with the following exceptions:

- The LSC update list access code is used
- The perform LSC call access code is used
- All long-list speed calls are completed after the LSC index number is dialed

The following parameters apply to the Speed Calling feature:

- LSC cannot be assigned to a station that has been assigned the SSC, Group Speed Calling, or Group Speed Calling Controller options.
- Each number can contain up to 30 digits.
- To include a DOD DN in an EBS speed call list, the list must include the DOD access code as part of the speed call digit string.

## Group Speed Calling and Group Speed Calling Controller (GSC and GSCC)

These two interrelated features, GSC and GSCC, enable an EBS speed call group to maintain and use a Group Speed Calling list, which associates each of up to 30 frequently called numbers with a two-digit Group Speed Call index number (any digit from 20 through 49). A maximum of 20 speed call groups can exist within one EBS customer group.

The GSC feature allows an EBS user to call a GSC list DN by dialing the access code followed by the DNs two-digit GSC index number. The frequently called number is then automatically dialed. Each number can consist of up to 30 digits.

The GSCC feature, which can be assigned to one or more stations in an EBS speed call group, allows the GSC list to be established and maintained. A user at a designated GSCC station may enter or change a number in the GSC list by performing the following actions:

**1)** Go off-hook, receive regular dial tone, and dial the GSC feature code.

**2)** After receiving special dial tone, dial the two-digit GSC index number to be entered or changed.

**3)** Dial the frequently-called number to be associated with the GSC index number entered in Step 2.

**4)** An interdigital timeout will occur (see ***Note***). If there has been no dialing error (see the paragraph following the ***Note***), confirmation tone is given after the entry is complete, and the list is then updated.

***Note:*** *The interdigital timing for GSC list updates is as follows:*

- After collecting the GSC code, a long interdigital timeout
- Until seven GSC entry digits are received, a long interdigital timeout
- After seven GSC entry digits are received, a short interdigital timeout

The following treatments are given to dialing errors that may occur during GSC updates:

- If the user dials an invalid GSC index number, an asterisk (*), or more than the allowed number of digits (15 or 20), the call is generically routed to dialing error, and the list is not updated.
- If the user goes on-hook before receiving the confirmation tone, the list is not updated.

The following parameters apply to the GSC and GSCC features:

- The GSC and GSCC features are assigned on an EBS station basis in overlay DN. This overlay allows the assignment, deletion, and query of the GSC features. When these station options are assigned, the EBS speed call group number (01 through 20) is also assigned.
- Each line within a speed call group may have an individual SSC list, but cannot have an individual LSC list.
- Each number in an EBS speed call list can contain up to 15 digits, up to 20 digits if Equal Access Abbreviated Dial 3 is configured, or up to 24 digits for Generic 404.20 and later 404-Series generics.
- To include a DOD DN in an EBS speed call list, the list must include the DOD access code as part of the speed call digit string.

## Call Hold (CHD)

The EBS CHD feature is identical to IBS Call Hold. Refer to “Call Hold” under “Integrated Business Services.”

## Busy Transfer (BTF)

The EBS BTF feature is identical to IBS Busy Transfer. Refer to “Busy Transfer” under “Integrated Business Services.” ISDN subscribers can redirect voice information calls that terminate to busy ISDN EBS members. All three Busy Transfer features, BTF, BTFA and BTFI support ISDN.

Only Voice and Circuit Mode Data calls from outside the EBS group, coming in to the EBS group, are redirected from the busy ISDN BTF subscriber. All other calls terminating to a busy ISDN EBS member will receive busy treatment. All terminating Voice or Circuit Mode Data calls are redirected from a busy ISDN BTFA subscriber. Only intragroup Voice or Circuit Mode Data calls are redirected from a busy ISDN BTFI subscriber. All other calls terminating to a busy ISDN EBS member receive busy treatment.

## Busy Transfer All (BTFA)

The EBS BTF feature is identical to IBS Busy Transfer. Refer to “Busy Transfer” under “Integrated Business Services.”

## Busy Transfer Intragroup (BTFI)

The EBS BTF feature is identical to IBS Busy Transfer. Refer to “Busy Transfer Intragroup” under “Integrated Business Services.”

## User Transfer (UTF)

The EBS UTF feature is identical to IBS User Transfer, except for the availability of the CTO feature listed below. Refer to “User Transfer” under “Integrated Business Services.”

For 500/2500 type terminal, an EBS user can transfer an established call to two subscribers by performing a switch-hook flash, dialing the third party to which the call is to be transferred, and performing a disconnect at any time following completion of dialing.

For Meridian Business Sets, the EBS user first presses the *Transfer* key and dials the third party to which the call is to be transferred, and performing a disconnect at any time following completion of dialing.

For SIP lines, the call transfer is performed using instructions for the particular SIP client device.

## Three-Way Calling (3WC)

The EBS 3WC feature is identical to IBS Three-Way Calling. Refer to “Three-Way Conference Calling” under “Integrated Business Services.”

## Call Transfer Outside (CTO)

The CTO feature, an enhancement to user transfer (UTF), is assigned on a per-EBS-group basis. CTO enables an EBS user to transfer an established call to two subscribers outside of the EBS group.

The following conditions apply to the CTO feature:

- Except for SIP subscribers, the EBS user must be assigned the UTF station option (see overlay DN (STN) in NTP 297-3601-311, *Data Modification Manual*). For SIP lines, UTF is not compatible. If the EBS group has CTO enabled, all SIP lines in the EBS group receive the CTO capability.
- The EBS group to which the user belongs must be configured with the CTO feature (see overlay HUNT (EBS) in NTP 297-3601-311, *Data Modification Manual*).

## Call Forwarding (CFW)

The EBS CFW feature is identical to the Custom Calling Services Call Forwarding feature except for the parameter listed below. Refer to “Call Forwarding” under “Custom Calling Services.”

In addition, with the EBS CFW feature, calls can be forwarded to DNs within or outside the EBS group. When CFW is activated to a DN outside the EBS group, a second dial tone will be returned to the user when the DOD access code is dialed.

### Call Forward Reminder Tone Disable

This feature determines whether a Meridian Business Set (MBS) subscriber receives a reminder tone when call forwarding takes place. When the feature is active, an incoming call is forwarded and no indication is given to the forwarding station. The feature can be applied only to the MBS Call Fwd key, through overlay MBS (MBS/MBST), prompt KTYP.

> ***Note:*** *This feature applies only to the Call Forwarding (CFW) feature.*

## Call Forwarding DMO Activation/Deactivation

The EBS Call Forwarding DMO Activation/Deactivation feature is identical to the Custom Calling Services feature.

## Call Forwarding Limitation

The EBS Call Forwarding Limitation feature is identical to the Custom Calling Services feature.

## User Programmable Call Forward Busy Don't Answer

The EBS User Programmable Call Forward Busy Don't Answer feature is identical to the Custom Calling Services feature.

## Usage Sensitive User Programmable Call Forward Busy Don't Answer

The EBS Usage Sensitive User Programmable Call Forward Busy Don't Answer feature is identical to the Custom Calling Services feature.

## Call Waiting (CWT)

The EBS CWT feature is identical to IBS Call Waiting. Refer to “Call Waiting” under “Integrated Business Services.”

## Call Waiting Chaining (CWC)

The EBS CWC feature is identical to IBS Call Waiting Chaining. Refer to “Call Waiting Chaining” under “Integrated Business Services.”

## Cancel Call Waiting (CCWT)

The CCWT feature allows a subscriber with the Call Waiting, Call Waiting Incoming, or Call Waiting Intragroup option to inhibit the call waiting tone for the duration of a single call. CCWT also allows a subscriber to inhibit the call waiting tone that is imposed by a call originator using the Call Waiting Origination or Dial Call Waiting option. Refer to the description of CCWT under “Custom Calling Services” for information on CCWT operation.

## Call Waiting Incoming (CWTI)

The CWTI feature provides call waiting only for calls from outside the EBS group. The feature informs an EBS subscriber who is busy on an existing two-way call that a third party from outside the EBS group is calling. Calls from within the EBS group receive a busy signal.

CWTI is part of the Call Waiting Enhancements feature package. The operation of call waiting from CWTI is the same as that for IBS Call Waiting. Refer to “Call Waiting” under “Integrated Business Services.”

The following parameters apply to the CWTI feature:

- If Cancel Call Waiting has been activated at the called station, call waiting from CWTI will not be performed.
- CWTI cannot be assigned to a station that has been assigned the Busy Transfer, Busy Transfer All, Busy Transfer Intragroup, or Inhibit Call Waiting options.

## Call Waiting Intragroup (CWIG)

The CWIG feature provides call waiting only for calls from inside the EBS group. The feature informs an EBS subscriber who is busy on an existing two-way call that a third party from inside the EBS group is calling. Calls from outside the EBS group receive a busy signal.

CWIG is part of the Call Waiting Enhancements feature package. The operation of call waiting from CWIG is the same as that for IBS Call Waiting. Refer to “Call Waiting” under “Integrated Business Services.”

The following parameters apply to the CWIG feature:

- If Cancel Call Waiting has been activated at the called station, call waiting from CWIG will not be performed.
- CWIG cannot be assigned to a station that has been assigned the Busy Transfer, Busy Transfer All, Busy Transfer Intragroup, or Inhibit Call Waiting options.

## Call Waiting Origination (CWTO)

The CWTO feature allows an EBS subscriber to impose call waiting on another subscriber who is in the same EBS group and does not have the CWT option assigned. If a call is made from a station with the CWTO option to a station in the same EBS group and the called station is in a talking state, call waiting will automatically be imposed on the called station. If the called party is not in a talking state, the call originator will get a busy tone and call waiting will not be imposed. If the called station is idle, the call will terminate normally.

CWTO is part of the Call Waiting Enhancements feature package. The operation of call waiting from CWTO is the same as that for IBS Call Waiting. Refer to “Call Waiting” under “Integrated Business Services.”

The following parameters apply to the CWTO feature:

- A SIP Gateway line can be assigned the CWTO option and may impose call waiting on another (non-SIP) member of the EBS group. However, CWT cannot be imposed on a SIP line by another member of the group since the SIP call-waiting function is controlled locally by the SIP client device.
- If Cancel Call Waiting has been activated or the Inhibit Call Waiting option has been assigned to the called station, Call Waiting from CWTO will not be performed and the caller will receive a busy tone.
- CWTO cannot be assigned to a station that has been assigned the Dial Call Waiting option.
- If Busy Transfer has been activated at the called station, a call will be transferred only if Call Waiting cannot be performed.
- If Call Waiting is imposed on a station that is busy and has Directory Number Hunting activated, directory number hunting will be performed before Call Waiting.

## Dial Call Waiting (DCWT)

The DCWT feature allows an EBS subscriber to impose call waiting on another subscriber who is in the same EBS group and does not have the CWT option assigned. DCWT is the same as the Call Waiting Origination feature except that the originator must use an access code to impose call waiting. To activate DCWT, the call originator must go off-hook, receive dial tone, dial the DCWT access code, receive a special dial tone, and dial the one to four Station-to-Station digits of the terminating directory number.

DCWT is part of the Call Waiting Enhancements feature package. The operation of call waiting from DCWT is the same as that for IBS Call Waiting. Refer to “Call Waiting” under “Integrated Business Services.”

The following parameters apply to the DCWT feature:

- A SIP Gateway line can be assigned the DCWT option and may impose call waiting on another (non-SIP) member of the EBS group. However, CWT cannot be imposed on a SIP line by another member of the group since the SIP call-waiting function is controlled locally by the SIP client device.

- If Cancel Call Waiting has been activated or the Inhibit Call Waiting option has been assigned to the called station, call waiting from DCWT will not be performed and the caller will receive a busy tone.
- DCWT cannot be assigned to a station that has been assigned the Call Waiting Origination option.
- If busy transfer has been activated at the called station, a call will be transferred only if call waiting cannot be performed.
- If call waiting is imposed on a station that is busy and has Directory Number Hunting activated, directory number hunting will be performed before call waiting.

### Inhibit Call Waiting (ICWT)

The ICWT feature allows an EBS subscriber to prevent call waiting from being imposed by a calling party with the Call Waiting Origination or Dial Call Waiting feature. The calling party will receive a busy tone if the called station is not idle.

ICWT is part of the Call Waiting Enhancements feature package.

The following parameter applies to the ICWT feature:

- ICWT cannot be assigned to a station that has been assigned the Call Waiting, Call Waiting Incoming, or Call Waiting Intragroup options.

### Distinctive Ringing (DSR)

The EBS DSR feature is identical to IBS Distinctive Ringing. Refer to “Distinctive Ringing” under “Integrated Business Services.”

### Ring Again (RAG)

The EBS RAG feature is identical to the Custom Calling Services feature.

The following conditions apply to the Ring Again Feature

- The originator and target must be served by the same switch.
- The originating station must have the RAG station option and the target station must not have the RAG denied station option.
- The originating station can only have one RAG request active at any time.
- The target station can not exceed 16 RAG requests, which are served in the order in which they are received.
- RAG is assigned on an IBS-station basis in Overlay DN.
- RAG cannot be invoked against a station that has performed BTF, BTFA, or BTFI.
- RAG re-ring is not modified when the receiving station has DSR.

- The originating station can not have passed through a virtual facility group in reaching the target.

## Ring Again Denied (RAGD)

The EBS RAGD feature is identical to the Custom Calling Services feature.

## Restricted Station Options (RES1, RES2, LOCO)

EBS stations can be fully/partially restricted through the use of special station options. Use of these options supports multi-level restrictions for both incoming and outgoing stations. The Restricted Station Options for outgoing calls (RES1 and RES2) allow selective screening on certain stations and can be used on stations with or without the EBS option. The Restricted Station Option for incoming calls (LOCO) only allows calls from members of the same EBS group. The Restricted Station Options feature is assigned through Overlay DN.

## Call Park

Call Park enables members of an EBS group to park a call against their DN or Group Intercom (GIC) member number and continue to originate and receive calls on that set or MBS key. The Call Park feature is available in two versions: *Call Park* and *Directed Call Park*. Either 500/2500 telephone sets, M5000-Series business telephone sets or SIP devices may be used with this feature.

*Call Park* is used mainly when the location of the called party is unknown. Subscribers who have the Call Park option assigned may park a DN or GIC call against their directory number until the target party is paged and the call is retrieved from that station or another station in the same EBS group.

To park either a DN or GIC call when using a 500/2500 set, the subscriber flashes the hookswitch and dials the Call Park feature access code. The subscriber is disconnected from the call, given confirmation tone followed by reorder tone, and can then originate or receive calls.

To park either a DN or GIC call when using an M5000-Series set, the subscriber presses the CALL PARK key, if the set is so equipped. The key lights, the subscriber is disconnected from the call and is given confirmation tone, followed by silence. The CALL PARK key is extinguished and the subscriber is then able to originate or receive calls. If the M5000-Series set is not equipped with a CALL PARK key, the subscriber can park a DN call by pressing the CONF3 or UTF key, which lights, and dialing a Call Park access code after receiving a special dial tone. The CONF3/UTF key is then extinguished, the subscriber is then disconnected from the call and is given confirmation tone, followed by silence. After the call is parked, the parked party (calling party) hears either ringback or silence as designated for the EBS group. On an M5000-Series set, a GIC call is parked against the GIC member number of the GIC key; a DN call is parked against the directory number of the DN key.

To park a call when using a SIP device, the subscriber either flashes the hookswitch (500/2500 set off of the SIP device) or accesses another line key and dials the Call Park feature access code. The subscriber is disconnected from the call, given confirmation tone followed by reorder tone, and can then originate or receive calls.

*Directed Call Park* is used when the location of the called party is known and enables subscribers to park a call against the DN or GIC member number of the called party.

To park the call using the Directed Call Park feature when using a 500/2500 set, the subscriber flashes the hookswitch and dials the Directed Call Park feature access code followed either by the station-to-station (STS) code of the target station if the call is to be parked against a DN or the GIC access code and GIC member number if the call is to be parked against a GIC member. The parking party is disconnected from the call, given confirmation tone followed by silence, and can then originate or receive calls.

To park either a DN or GIC call using an M5000-Series set, the subscriber presses the DIRECTED CALL PARK key, if the set is so equipped. The key lights, and after receiving special dial tone, the subscriber dials either the STS code of the target station if the call is active on a DN key or the GIC member code if the call is active on a GIC key. The subscriber is then disconnected from the call and is given confirmation tone, followed by silence. The DIRECTED CALL PARK key is extinguished and the subscriber is then able to originate or receive calls. If the M5000-Series set is not equipped with a DIRECTED CALL PARK key, the subscriber can park a DN call by pressing the CONF3 or UTF key, which lights, and dialing a Directed Call Park access code after receiving a special dial tone. The CONF3/UTF key is then extinguished and the subscriber dials the STS code. The subscriber is then disconnected from the call and is given confirmation tone, followed by silence. After the call is parked, the parked party (calling party) hears either ringback or silence as designated for the EBS group.

To park the call using the Directed Call Park feature when using a SIP device, the subscriber either flashes the hookswitch (500/2500 set off of the SIP device) or accesses another line key and dials the Directed Call Park feature access code followed either by the station-to-station (STS) code of the target station if the call is to be parked against a DN. The parking party is disconnected from the call, given confirmation tone followed by silence, and can then originate or receive calls.

A time limit for parked call retrieval, from 12 to 360 seconds, may be set. If the time limit expires before the call is retrieved, the call is routed back to the subscriber who parked the call. The subscriber using a 500/2500 set is alerted either by distinctive ringing or normal ringing. The subscriber using an M5000-Series set is alerted by a combination of flashing CALL PARK/DIRECTED CALL PARK key, ringing (or buzz tone), and the display of a “CALL PARK RECALL” message, if the reason display (RDSP) option for the EBS group is set to “yes.” If the subscriber's station is busy, the recall timer is reset up to three times. After the third recall attempt fails, the call is forwarded to a pre-designated number, if the number is assigned. If the number is not assigned, the call remains parked either until if is retrieved, or the parked party releases from the call. If the timer is set at 0, the call remains parked until either the called party answers the call or the call is abandoned by the calling party.; call forwarding does not apply.

To retrieve a parked call, the called party may use a station that is a member of the EBS group. A subscriber using a 500/2500 set goes off-hook and dials the *park retrieve* access code, and after hearing special dial tone, dials the STS of the station against which the call is parked. If the call was parked against a GIC member, the subscriber first dials the *park retrieve* access code, and after hearing special dial tone dials the GIC access code, and then after hearing special dial tone again dials the GIC member number. A subscriber using an M5000-Series set presses the DN key, and after hearing special dial tone presses the CALL PARK/DIRECTED CALL PARK key, and then after hearing special dial tone again, dials the STS code of the station against which the call is parked. If the call is parked against a GIC member, the subscriber presses the GIC key, and after hearing special dial tone presses the CALL PARK/DIRECTED CALL PARK key, and after hearing special dial tone again, dials the GIC member number. If the M5000-Series station is not equipped with the CALL PARK/DIRECTED CALL PARK key, the subscriber presses a DN key, dials the *park retrieve* access code, and retrieves the call in the same manner as described above for a subscriber using a 500/2500 set. If the call was parked against a GIC member, only the CALL PARK/DIRECTED CALL PARK key will operate.

### Conditions applying to Call Park

The following conditions apply to the Call Park feature:

- Calls can be parked against a station that is a member of a directory Line Hunt group, is assigned the SHU option, and whose key is activated.
- Ring Again requests will ring the called party when both parties are in idle state even when the called station has a call parked against it.
- If the value of the parked call retrieval time limit is set at zero, a call remains parked either until it is retrieved or until the parked calling party disconnects.
- A subscriber using an M5000-Series set can retrieve a call while in talking state, by pressing the CONF3 or UTF key.

### Call Park and Directed Call Park for ISDN subscribers

Beginning in Generic 411.10, the Call Park and Directed Call Park features, integrated into a single feature called Integrated Call Park (I-PRK), are available to non-PCS ISDN BRI lines. I-PRK enables ISDN users who subscribe to Enhanced Business Services (EBS) to park voice-type calls on a directory number, from which they may be retrieved by the parking party or a member of the parking party's group.

A “IPRK” feature activator key assigned in a terminal configuration (TCGN) is used to park calls. A PRKR feature indicator can also be assigned and can be associated with each DNCT appearance. Calls parked against the DNCT appearance then cause the associated PRKR feature indicator to light. Several feature activators can be lit simultaneously.

To retrieve a call, an ISDN subscriber presses a DN key and, after hearing special dial tone, presses the PRKR feature activator associated with a lighted PRKR feature indicator. One PRKR feature activator can be assigned for each DNCT appearance. After hearing special dial tone again, the subscriber presses the pound sign (#) key to accept the STS code of the associated DNCT. The subscriber can, instead, dial an STS code to override the default STS code. To retrieve a call parked against another EBS member, the subscriber presses the appropriate DN key and then a PRKR feature activator, and then dials the EBS member's STS code. The ISDN subscriber can also retrieve a call by dialing the Call Park retrieval access code followed by an STS code, when a PRKR key is not assigned.

The Call Park recall process and call forwarding applies both to ISDN and non-ISDN subscribers.

ISDN terminals can park and retrieve calls using EBS group station-to-station (STS) numbers. However, because Group Intercom (GIC) dialing is not supported by ISDN, ISDN terminals cannot park or retrieve calls using GIC member codes.

## Camp-On

The Camp-On feature enables a member of an EBS group to use the Call Transfer feature to extend a call originating either from inside or outside of the group to a busy station. Either 500/2500 telephone sets or M5000-Series business telephone sets may be used with this feature.

### Three-party Camp-On calls

Normally, the Camp-On feature is used when a caller doesn't know the number of the party being called and dials an administration number. The administrator receiving the call then initiates a three-way call to the requested party. If the administrator is using a 500/2500 telephone set, the three-way call is made by flashing the hookswitch and dialing the number of the requested party. If the administrator is using an M5000-Series telephone set, the three-way call is made by pressing the UTF key and dialing the number of the requested party. If the target party's station is busy, and if Camp-On is allowed, the administrator is given Camp-On allowed audio/visual treatment. An administrator using a 500/2500 set hears a confirmation tone; at an M5000-Series set, the administrator sees the CAMP key light, the “CAMP-ON” message display (if the RDSP, reason display, option is set to “yes” for the EBS group), and hears a confirmation tone. The target party hears a *call waiting* tone and, on MBS sets so equipped, sees the UTF key wink. If Camp-On cannot be performed on a target station, the administrator is given busy signal. When the party at the busy station subsequently goes on hook, the station automatically rings and can be connected to the waiting call.

When the call is camped-on, the administrator can then reconnect to the calling party by flashing the hookswitch on a 500/2500 set, or by pressing the UTF key on an M5000-Series business set. If the calling party chooses not to wait for the target party to answer, the administrator can then cancel the camped-on call by performing a hook-flash at a 500/2500 set, or by pressing the UTF key at an M5000-Series set. If the calling party chooses to wait for the target party to answer, the administrator disconnects from the camped-on call by going on-hook at a 500/2500 set, or by pressing the REL key at an M5000-Series set. The calling party is then camped-on to the target party's station and receives a specified Camp-On audio treatment (either ringback or quiet tone). When disconnected from a camped-on call, the administrator's station is free to receive and camp-on other calls.

A time limit for answering the call, from 12 to 360 seconds long, may be set. If the busy station with the camped-on call goes on-hook within this time limit, the station rings and the camped-on call can be answered. If the time limit expires before the camped-on call is answered, the administrator's 500/2500 station is rung with standard or distinctive ringing, or the CAMP key lights and the “CAMP RECALL” message displays (if the RDSP, reason display, option is set to “yes” for the EBS group) on the administrator's M5000-Series set. If the administrator's set is busy, the recall timer is restarted. If the administrator doesn't answer within the recall ring time set up for the EBS group, the recall timer is restarted. When the administrator answers the re-call, the camped-on call is cancelled and the administrator is reconnected with the original calling party. While the administrator's station is being recalled, the target party can still answer the camped-on call. If the target party answers the camped-on call, the recall of the administrator's station is cancelled.

### Two-party Camp-On calls

When the Camp-On feature is used with only two parties, the calling party camps on directly to the target party's station. If the calling party disconnects before the camped-on call is answered, the Camp-On is cancelled. If the target party answers the camped-on call, a connection is set up between the calling and target parties.

### Conditions applying to Camp-On

The following conditions apply to the Camp-On feature:

- Camp-On cannot be performed on lines that have a previously camped-on call, a call on hold, a call waiting, or are the controlling station in a Three-Way call.
- Camp-On is not allowed on lines that are hunt-group members.
- If the value of the camped-on call retrieval timer is set at zero, a call remains camped on either until it is answered or until the camped-on calling party disconnects.
- Stations with Call Waiting cannot have camped-on calls.

## Virtual Facilities Group Controls (VFGC)

Virtual Facilities Group Controls (VFGC) is an enhancement to the EBS feature package and is used to control the use of network resources with virtual facility groups (VFG). A VFG is a logical trunk group which limits the number of simultaneous incoming or outgoing EBS calls.

The three functional types of VFGs, assigned as EBS customer group options and administered in overlay Hunt, are incoming, outgoing, and two-way.

Originating VFGs are Direct Outward Dial (DOD), Outgoing Wide Area Telephone Service (OWTS) and variable. The DOD VFGs are provided for EBS subscribers without an outwats station option, while the OWTS VFGs are provided for EBS subscribers with an outwats station option. A subscriber will pass through either a DOD VFG, an OWTS VFG, or a variable VFG when originating a call. There are five variable VFGs that can be used for any originating purpose.

Incoming VFGs are Direct Inward Dial (DID) and Incoming Wide Area Telephone Service (IWTS). The DID VFG is provided for EBS subscribers without an inwats station option, while the IWTS VFG is provided for EBS subscribers with an inwats station option. A subscriber will pass through either a DID VFG or an IWTS VFG when receiving an incoming EBS call.

Both DID and DOD traffic may have one-way and two-way VFGs assigned concurrently. When this occurs, it is necessary to assign one as the primary VFG, and the other as the overflow VFG.

A virtual facility will remain associated with a call and considered unavailable until either the call is disconnected, or, the call is routed to a generic condition.

The following conditions apply to VFGC:

- A maximum of six VFGs can be associated with any given call.
- A VFG is associated with the call, not with the EBS subscriber.
- When the VFG limit is reduced for an existing VFG, the new limit becomes effective when the number of active calls in that VFG drops to that level.
- Only one INWATS VFG per EBS group is available.

### Virtual Facilities Group Control (VFGC) enhancements for Group INWATS and Group OUTWATS

The enhancements to VFGC provide Group INWATS (GIWT) and Group OUTWATS (GOWT) options. This enhancement allows multiple INWATS and OUTWATS groups to be defined within an EBS group. The number of VFGs available to an EBS group is increased by a total of 32 per EBS group (16 for GOWT and 16 for GIWT).

The GIWT option provides the telco the ability to control the number of calls to the EBS group. Each EBS group can have a maximum of 16 INWATS groups defined. Each INWATS sub-group is assigned a separate INWATS VFG, and each INWATS VFG can be a different size. EBS members are assigned to INWATS groups through the GIWT station option. This is an enhancement to VFGC in which all calls to stations with the INWATS station option (IWT) passed through a single INWATS VFG. By default, both local and trunk calls are allowed to terminate at a GIWT station. An option is provided on a per EBS group basis to limit only trunk calls to terminate at a GIWT station.

The GOWT option provides members of an EBS group the ability to make WATS calls without assigning each member the OUTWATS station option. The user is required to dial an access code, or codes, defined by the telco, to gain access to particular OUTWATS facilities. Each EBS group can have a maximum of 16 group OUTWATS facilities defined. Each group OUTWATS VFG will have a WATS band number and WATS service type assigned, and each OUTWATS VFG can be a different size.

By default, each member in the same EBS group has access to the GOWT VFGs; however, individual stations can be restricted by assigning the Group OUTWATS Denied (GWTD) station option. Three levels of OUTWATS restriction can be defined within each EBS group through EBSP translator and GWTD test. This provides the flexibility to selectively deny access to the group OUTWATS facilities. An EBS station can be denied access to any one of the three levels (1, 2, 3) of OUTWATS restriction or any combination of the levels.

The conditions listed for VFGC apply to the group WATS VFGs. In addition, the following conditions apply:

- Seizing a group OUTWATS VFG does not ensure call completion. The subscriber must dial within the prescribed calling areas associated with the group outwats VFG that was seized.
- Automatic overflow between group WATS resources is not provided. Overflow of the Group INWATS resources is not provided. Overflow between the Group OUTWATS resources can be accomplished in translations.
- The translations test for OUTWATS denial should be defined before the VFG OUTWATS test is defined so that a VFG resource is not assigned to a call attempt that may be blocked by the OUTWATS denial option.
- INWATS calls to stations with the group INWATS station option will not complete if the associated Group INWATS VFG is not defined. Non-INWATS calls to stations with the group INWATS station option will attempt to complete and may pass through the DID VFG, if defined.
- OPM printouts for the EBS groups will only include VFG peg, block, and usage counts for the VFGs that are defined for that EBS group. The VFGs that are not defined will not be included in the printout. All VFGs will be included in the EADAS printout.

### VFG Special Billing Number

The VFG Special Billing Number feature provides the EBS administrator with the capability of adding a special billing number (SBN) to each VFG type assigned to a Centrex group. Because the SBN is charged for the VFG subscribers' calls, the Centrex administrator is provided a telephone bill with calls consolidated according to each VFG type assigned to the Centrex group.

The VFG Special Billing Number feature changes only the identification of the calling number to bill, not the manner in which charges are calculated. With the VFG Special Billing Number feature implemented, one directory number (the SBN) is billed for all calls in the group (INWATS, OUTWATS, Group INWATS, Group OUTWATS) rather than each station being billed separately.

An SBN can be located either in the central office or in another exchange. Although an SBN is normally a member of the VFG group, it is not mandatory. The SBN is assigned in Overlay HUNT (EBS), prompt “SBN.”

The following conditions pertain to the VFG Special Billing Number feature:

- Charges are not incurred for the initial leg of a call that terminates to IWT and GIWT stations after a call transfer.

- When a mismatch in class-of-service (COS) occurs between the calling party and the SBN, the COS of the calling number is used to determine the charging applicable to the call.
- If the VFG Special Billing Number and Special Party Billing (SPB) features are both assigned to a VFG, the Special Billing Number is charged rather than the Special Party Billing number.
- When an outgoing call is originated by a VFG member while the VFG Special Billing Number feature is active, the response to any automatic number identification (ANI) request is the SBN.
- When a call terminates at a station with Calling Number Delivery (CND), the number identified with the call is that of the calling party and not the SBN associated with the calling party's VFG. When a VFG group member originates a call while the VFG Special Billing Number feature is active, the calling number identified in the IAM message is the calling number and not the SBN. If CND blocking is active during the call, the SBN is not substituted for the calling number.
- A maximum of six VFGs can be attached to a single call. Because only one SBN can be billed for a call, the SBN (if any) of the most recently attached VFGs is billed.

## Message Detail Recording (MDR)

Message Detail Recording provides detailed information for calls originating at stations in EBS groups. The ability to record MDR information for all incoming non-intra-group calls or only incoming trunk calls is provided. The MDR information is recorded by enabling the MDR call types through data modification. The collected information is sent to the EBS group by way of a data link to customer provided equipment (CPE), or to a revenue accounting office (RAO), which sends the information to the EBS group.

When the MDR data is sent to a data collector (a third party manufacturer's device), by way of an available port on either a Serial Data Interface pack (NT3T09) or a Dual Serial Data Interface pack (NT3T80), the MDR data is formatted using the Bellcore Customer Premise (CP) MDR format. The MDR data is formatted and sent to a CP data collector where the customer can retrieve the information by way of the EBS customer's own CPE. MDR data sent to the CP data collector does not contain telco proprietary data.

When the MDR data is sent to the RAO, the MDR data is formatted using the Bellcore modular format. The MDR data is included in the AMA billing records and entered into the AMA tape, along with the AMA billing data. The RAO formats the MDR data, per EBS customer group, from the billing records and MDR modules, and sends the formatted data to the EBS group. The RAO is responsible for removing any telco proprietary data that should not be sent to the EBS group.

To assign this feature to an EBS group, see overlay Hunt, prompting sequence EBS, in the NTP entitled *Data Modification Manual* (297-3601-311).

The following conditions apply to MDR:

- There is no limit to the number of EBS groups or EBS stations which may have MDR data recorded using Bellcore modular format.
- The Bellcore modular format (RAO) is available in the HSO, SSO, and LCC.
- The Bellcore modular format (RAO) requires the office to be using Bellcore AMA format.
- If MDR is provided in an SSO using the Bellcore CP format, the SSO must have its own serial data interface and data collector.
- The telephone company can specify whether the following types of calls receive MDR treatment: calls originating from the MDR entity; calls terminating to the MDR entity; both call attempts and completions involving the MDR entity, or only call completions involving the MDR entity; calls originating from the MDR entity that access the public network; intra-Centrex calls.
- The *no MDR* station option (NMDR) can be used to prevent billing on a station-by-station basis.

### Message Detail Recording enhancements

An enhanced version of MDR ensures that the DMS-10 switch processes MDR records in the manner prescribed in Bellcore documents, GR-610-CORE, Issue 1, and GR-1100-CORE, Issue 1. The details of the enhancements are described in the following paragraphs.

**MDR-CP with RAO backup** The DMS-10 switch can also send the MDR data directly to the revenue accounting office (RAO) if this third-party equipment is unavailable. Because the DMS-10 switch cannot really determine the status of a customer's CP equipment, the switch sends MDR Records to the RAO if it determines that the serial data interface to the customer's CP equipment is down.

**Creation of MDR-CP records with RAO format** A copy of a call's record is appended to related MDR modules and is then sent to the customer-provided equipment. The telco must update their customer premises software in order to accept BAF MDR-CP records from the DMS-10 switch.

**Digits Dialed Module 101** When the actual number to which a call is routed differs from the number dialed by the originator of the call, the actual digits received on the originating facility are recorded in Module 101. Certain features such as Automatic Recall record the originally-dialed number rather than the access code to the feature.

**Message Detail Recording Module 105** This module is always included in the MDR data record to carry the identity of the MDR customer to whom this data is being transmitted. It also identifies the type of originating and terminating facility as seen from the perspective of the switch creating the MDR record. The customer identity number is a unique number, up to 10 digits in length, defined in Overlay HUNT (EBS) (see NTP 297-3601-311, *Data Modification Manual*).

**Facility Identification Module 106** Module 104 is normally used for non-MDR purposes to record the identity of trunk facilities in AMA data records. For MDR records, however, the trunk identity provided in Module 104 is suppressed. Thus, Module 106 is used to record the private or virtual facility identification of calls that use the MDR customer's private or virtual facilities. Because it is possible that both the originating and the terminating facility are part of the MDR customer's private network, this module may be recorded twice in the MDR data record to identify the facilities involved in the call. Trunk data is not reported in the module since it is proprietary telco data.

**Business Features Module 107** This module contains information relating to the customer's private network and special features. If MDR subscribers' calls use any features that access their private networks, or if the calls use any special business features, then Module 107 is appended to the MDR record for those calls. Module 107 is applicable only for the DMS-10 Call Forwarding feature and toll diversion features.

**Data Suppression** MDR-CP billing records contain data that is proprietary to the telephone company handling a call. Since this data may divulge network configurations, troubles, and other proprietary information, before an MDR record containing such information can be sent to an MDR-CP customer, this proprietary data must be suppressed. Thus, in MDR-CP billing records, this kind of data is overwritten with hexadecimal “F” characters. Tables 3-A and 3-B list the data fields (data tables), which are described in NTP 297-3601-124, *Automatic Message Accounting System*, that are always suppressed.

| **Table 3-A: Data Tables Always Suppressed in MDR-CP Records** | |
|---|---|
| **Table Name** | **Table Number** |
| Timing Indicator | 7 |
| Study Indicator | 8 |
| Service Observed, Traffic Sampled | 10 |
| Trunk Network Number (not created in the DMS-10 switch) | 20 |
| Terminating Company | 56 |
| IC/INC Call Event Status | 58 |
| ANI Indicator | 60 |
| Tariff Utility (not created in the DMS-10 switch) | 183 |
| Call Identifier Billing (not created in the DMS-10 switch) | 186 |
| Counter Reset (not created in the DMS-10 switch) | 193 |
| X.75 Interface Identifier (not created in the DMS-10 switch) | 194 |
| Incoming/Outgoing Trunk Identification | 244 |

| **Table 3-B: Suppressed Public Facility MDR-CP Data Tables** | |
|---|---|
| **Table Name** | **Table Number** |
| Carrier Connect Date | 6 |
| Carrier Connect Time | 18 |
| Carrier Elapsed Time | 19 |
| Circuit Time (not created in the DMS-10 switch) | 26 |
| Routing Indicator | 59 |
| Trunk Group Number | 83 |

Any fields populated with a private (anonymous) number are also suppressed. For example, the CLASS features Automatic Callback and Automatic Recall can result in the recording of private numbers. In this event, the Study Indicator (data table 8) or the Called DN Descriptor module (Module 068) are used to indicate that a private number is in the terminating number fields (Tables 15, 16, and 17). If character 5 of the Study Indicator is set to 3 (“called directory number marked anonymous”), then the corresponding terminating number is suppressed in MDR-CP records.

**Study Records** The DMS-10 switch never sends AMA records for call type 067 (originating study record) or call type 036 (terminating study record) to a customer's premises. The DMS-10 switch sends, instead, call type 159 (message detail recording) to the customer's premises and call type 067 or 036 (as appropriate) to the RAO.

If an MDR customer requires MDR information for a particular call, but an AMA record would not normally be generated for that call, the DMS-10 switch creates an AMA record with call type code 159 (Message Detail Recording Data). This type of record reports usage for calls to an MDR customer when an AMA record is not otherwise applicable.

### Message Detail Recording FXD Customer Premise (CP) Billing

This feature provides fixed length MDR Bellcore AMA Format (BAF) Customer Premise records for operating companies that could before only process fixed length DMS-10 MDR CP records.

The MDR FXD CP format always records AMA call structure 0360 in the fixed length billing record. Additional information required is supplied in appended modules 101 (digits dialed), 105 (MDR), and the final module, 000. The billing records each contain 252 RS-232 characters. Each record begins with the ASCII characters, 007E0000.

The following conditions apply to the MDR FXD CP format:

- The MDR FXD CP format can contain information that is normally suppressed when sent to the subscribers' premises, including the trunk number and the digits dialed by the caller.
- MDR FXD CP billing does not alter billing record structures or content sent to the Regional Accounting Office (RAO).
- Variable-length BAF MDR records are the system default. Thus, the operating company must specify fixed-length format through the FRMT prompt in the Overlay CNFG (SYS) prompting sequence.

## Music on Hold

Music on Hold automatically provides music or announcements to calls placed on hold by the Camp On, Call Hold, Call Park, Directed Call Park, MBS Hold, ISDN Call Park, and ISDN Hold features. A caller placed on hold hears the music (or announcements) until either the caller hangs up or the DMS-10 switch takes the call off hold.

The music or announcement that the caller hears is acquired from an external source such as a radio station or a CD player over a dedicated analog or digital trunk. The DMS-10 switch can support up to 256 Music on Hold source trunks. A single source can supply music for any combination of EBS groups and EBS subscribers.

The following conditions apply to the Music on Hold feature:

- Music on Hold is provided on a flat-rate billing basis only.
- Access to the Music on Hold feature is through feature bit (see prompt MOH in Overlay CNFG (FEAT) prompting sequence).

- The music source used by the telco must have an output interface circuit designed for use with standard impedance telephony trunks; the music source can be physically located according to the telco's requirements.
- Fidelity of sound frequencies below 4000 Hz will be superior to sound frequencies above 4000 Hz.
- Music source trunks are not permitted on remote switches.
- A DMS-10 switch provides the ability to assign different music sources to different members of the same EBS group.

## EBS Group Name and Number

The EBS Group Name and Number feature enables a valid 10-digit directory number (DN) to be assigned to an EBS group through prompt GPDN of the overlay HUNT (EBS) prompting sequence. This 10-digit group DN and any associated name defined in the calling name database are delivered to the called party when any member of the EBS group originates a call to a party outside the business group. The individual member’s name and number continue to be delivered to the called party for intra-group calls.

Delivery of the EBS group name and number is always public for calls outside the EBS group and overrides all forms of suppression options at the DN level. However, system parameters for originating and terminating office-wide calling number delivery as well as originating and terminating office-wide calling name delivery suppression still determine the actual display status of the group name and number. (For an explanation of these system parameters, see prompts OSUP, TSUP, ONAS, and TNAS of the overlay CNFG (SYS) prompting sequence in NTP 297-3601-311, *Data Modification Manual*.)

## Meridian Business Sets

DMS-10 Meridian Digital Centrex (MDC) enables the Telco's business customer who subscribes to Enhanced Business Services (EBS) to integrate separate access lines into a single communications group. The Meridian Business Sets feature enables this MDC feature package to be used with Nortel M5000-Series business sets.

***Note:*** *The Multiline Variety Package (MVP) feature package was formerly known as Integrated Business Services. Integrated Business Services (IBS) is the name under which the Enhanced Business Services (EBS) feature package is now marketed. Thus, the terms IBS and EBS and the mnemonics for IBS and EBS features are used throughout the NTPs and in the DMS-10 software.*

A DMS-10 switch configured with the EBS feature package can support a single business group consisting of up to 3000 lines or up to 512 business groups with a total combined number of lines not exceeding 3000. Each line can be a member of only one business group. Although the MBS feature enables EBS service to be offered with Nortel M5000-Series business sets, Nortel 500/2500-type sets can still be used and intermixed with the M5000-Series sets. For a procedure used to set up basic MBS capabilities see SOP 0133 in NTP 297-3601-311, *Data Modification Manual*.

## M5000-Series Meridian Business Sets

Eight M5000-Series sets can be used with the MBS feature: M5008, M5208, M5009, M5209, M5112, M5312, M5216, and M5316.

### M5008 business set

The M5008 business set has the following features:

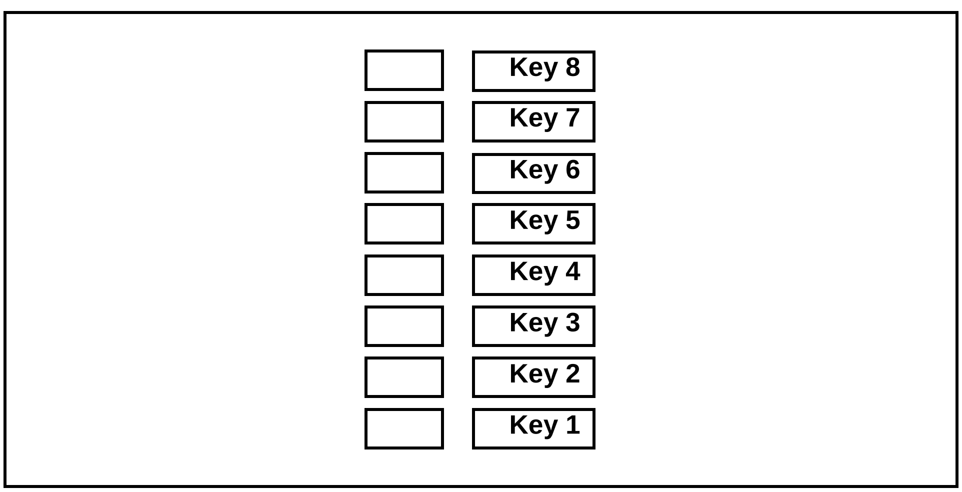
- 3 fixed keys without Liquid Crystal Display (LCD) indicators (release; hold; volume)
- dial pad consisting of 12 fixed keys
- 8 programmable feature/directory number keys with LCD indicators, shown in Figure 3-2.
- speaker for alerting tones, on-hook dialing, and intercom

### M5208 business set

The M5208 business set has the following features:

- 3 fixed keys without LCD indicators (release; hold; volume)
- dial pad consisting of 12 fixed keys
- 8 programmable feature/directory number keys with LCD indicators, shown in Figure 3-2
- speaker for alerting tones, on-hook dialing, and intercom
- alphanumeric display

**Figure 3-2: M5008 and M5208 programmable key configuration**



### M5009 business set

The M5009 business set has the following features:

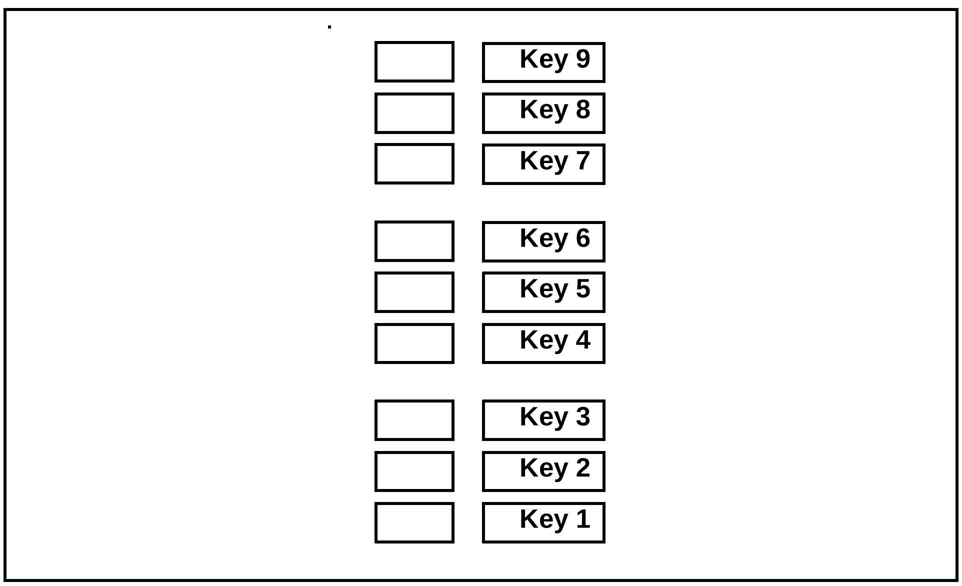
- 3 fixed keys without LCD indicators (release; hold; volume)
- dial pad consisting of 12 fixed keys
- 8 programmable feature/directory number keys with Liquid Crystal Display (LCD) indicators, shown in Figure 3-3
- 1 programmable feature key (key 9) without an LCD indicator, shown in Figure 3-3
- speaker for alerting tones, on-hook dialing, and intercom

### M5209 business set

The M5209 business set has the following features:

- 3 fixed keys without LCD indicators (release; hold; volume)
- dial pad consisting of 12 fixed keys
- 8 programmable feature/directory number keys with LCD indicators, shown in Figure 3-3
- 1 programmable feature key (key 9) without an LCD indicator, shown in Figure 3-3; this key can be used only for the Call Pickup feature
- speaker for alerting tones, on-hook dialing, and intercom
- alphanumeric display

**Figure 3-3: M5009 and M5209 programmable key configuration**



### M5112 business set

The M5112 business set has the following features:

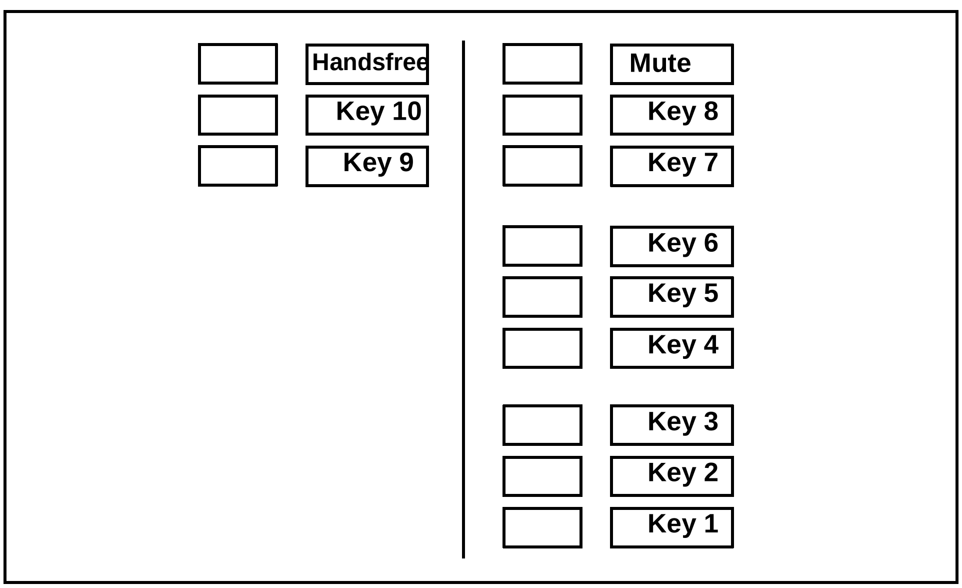
- 2 fixed feature keys with associated LCD indicators (hands-free and mute)
- 3 fixed feature keys without LCD indicators (release, hold, and volume)
- dial pad consisting of 12 fixed keys
- 10 programmable feature/directory number keys with LCD indicators, shown in Figure 3-4
- microphone and speaker for hands-free operation

### M5312 business set

The M5312 business set has the following features:

- 2 fixed feature keys with associated LCD indicators (hands-free and mute)
- 3 fixed feature keys without LCD indicators (release, hold, and volume)
- dial pad consisting of 12 fixed keys
- 10 programmable feature/directory number keys with LCD indicators, shown in Figure 3-4
- microphone and speaker for hands-free operation, alerting tones, and on-hook dialing
- alphanumeric display

**Figure 3-4: M5112 and M5312 programmable key configuration**



### M5216 business set

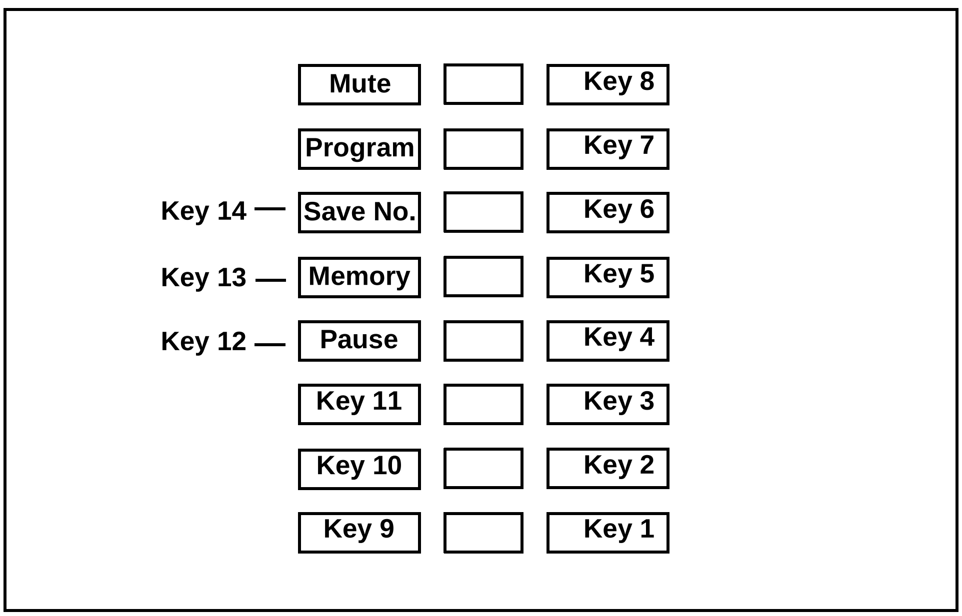
The M5216 business set has the following features:

- 1 fixed feature key with associated LCD indicators (handset mute)
- 3 fixed feature keys without LCD indicators (release, hold, and volume)
- dial pad consisting of 12 fixed keys
- 11 or 14 programmable feature/directory number keys with LCD indicators, shown in Figure 3-5

  ***Note:*** *The four keys supporting "local" features, Program, Save No., Memory, and Pause, are supported on the M5216 business set as set features. An M5216 set feature enables the Program key to be used to convert three of the keys (Save No., Memory, Pause) for use as central office-programmable keys.*

- speaker for on-hook dialing operation
- alphanumeric display

**Figure 3-5: M5216 programmable key configuration**



### M5316 business set

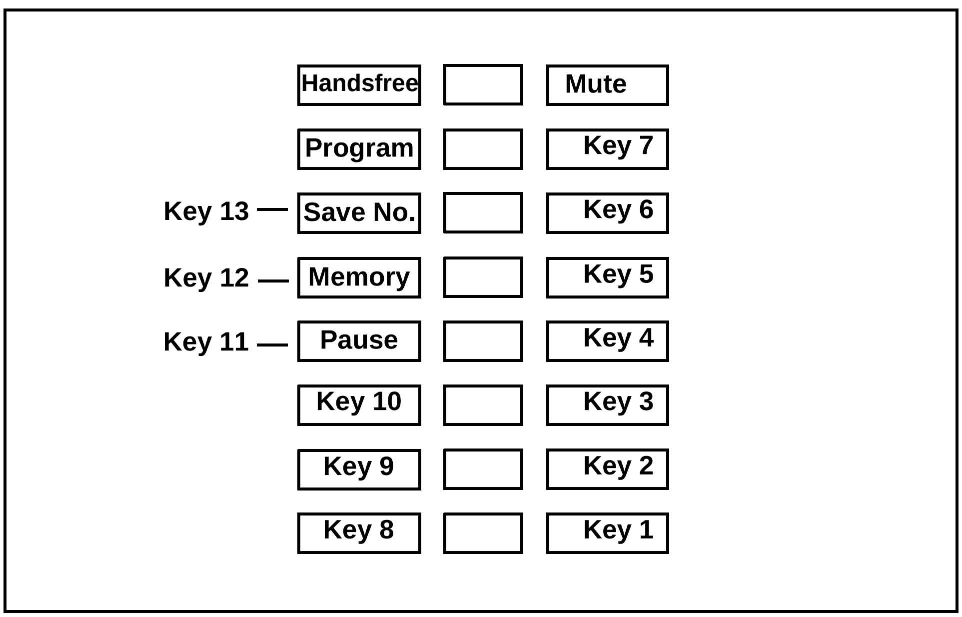
The M5316 business set has the following features:

- 2 fixed feature keys with associated LCD indicators (hands-free and mute)
- 3 fixed feature keys without LCD indicators (release, hold, and volume)
- dial pad consisting of 12 fixed keys
- 10 or 13 programmable feature/directory number keys with LCD indicators, shown in Figure 3-6

  ***Note:*** *The four keys supporting "local" features, Program, Save No., Memory, and Pause, are supported on the M5316 business set as set features. An M5316 set feature enables the Program key to be used to convert three of the keys (Save No., Memory, Pause) for use as central office-programmable keys.*

- microphone and speaker for hands-free operation, alerting tones, and on-hook dialing
- alphanumeric display

**Figure 3-6: M5316 programmable key configuration**



### M5000-Series business set key descriptions

*Fixed keys* are non-optional keys that are permanently labeled on the business set. The fixed keys on the M5000-Series business sets that are operational with the MBS feature include:

- dial pad; a standard dialing pad with 12 keys (0-9,*,#)
- hold key; allows the subscriber to place an active call on hold
- release key; terminates a completed or in-progress call
- volume control key; increases or decreases the volume of sound through the speaker or, when used in conjunction with the *Program* key, to scroll through business set features such as time and date display
- hands-free key; enables a subscriber to carry on a conversation with the handset in the cradle
- mute key; enables a subscriber to disable the microphone (handset transmitter) on the set during a call; the speaker remains on
- key 1 (lower right side of the set) must always be used as a DN key

The M5112, M5312, and M5316 models also have a hands-free key/LCD pair and a mute key/LCD pair. The LCD is always in one of four possible states: ON, OFF, FLASH (60 IPM - 1 s on, 1 s off), WINK (120 IPM - 750 ms on, 250 ms off).

Assignable keys can be set up either as directory number appearances or as feature appearances. Two types of DN appearances can be associated with an assignable key: primary DN (PDN) or secondary DN (SDN). A single PDN must be assigned on the business set; up to seven SDNs can also be assigned. The feature key appearances that may be assigned include:

- Group Intercom
- Ring Again
- Message Waiting
- Call Forwarding
- Call Park
- Directed Call Park
- Camp On
- Call Waiting
- Conf 3
- Pick up
- Speed Call
- Call Transfer
- Automatic Dial
- Direct Station Select
- Make Set Busy

### M5000-Series business set alphanumeric display

The M5208, M5209, M5216, M5316, and M5312 business sets are equipped with a 48-character LCD display. The display is arranged in two lines of 24 characters each. The display provides a subscriber with visual feedback during call origination and termination, telephone set programming, and feature activation operations. The lower display line normally echoes digits that are dialed, while the upper line displays call conditions.

Newer sets offer time and date on display, as set features. Displays show call information for:

- Call Transfer
- Call Pickup
- Call Park Recall
- Camp On Recall
- Call Forwarding

- Call Forwarding Busy
- Call Forward No Answer
- Three-Way Calling

### M5000-Series business set alerting and progression tones

A 500 Hz tone produced by an M5000-Series business set tone generator is used for off-hook alerting with features such as Call Waiting. All other alerting and progression tones (including ringing tone) are provided by the DMS-10 switch Tone and Digit Sender (NT4T01) pack or by Global Tone Services in the NT8T04 pack.

### M5000-Series business set hardware configuration

The M5000-Series business sets are connected to NT6X21AC line circuits by way of 2-wire non-loaded subscriber loops. Voice communication, call progress and alerting tones, signaling, and power are provided over the 2-wire connection. The transmission bandwidth of the subscriber loop is shared by a voice channel that uses the 300-3400 Hz band and a signaling channel that uses the 6-10 kHz band.

Signaling messages are passed across the signaling channel under a low level signaling protocol (W78) that governs message exchange. Transmissions consist of 16 bits: eight bits are used to convey command data (hook-switch status, digit/feature key depression, information concerning LCD control, etc.) and eight bits are used for timing reference, addressing, direction control, and message integrity.

The M5112, M5312, M5208, M5209, M5216, and M5316 business sets are powered by a 16V ac external power supply (plug-in transformer) to support the display, hands-free operation, and assignable-key LCDs. The power supply is supplied with the business set.

For information about alerting and call progression tones associated with the M5000-Series business sets, see NTP 297-3601-180, *System Performance Specifications*.

### M5000-Series business set call origination and termination

To originate a call, the subscriber lifts the handset or presses the *Handsfree* key. Assuming that automatic primary DN (PDN) selection is used, the PDN LCD lights, the subscriber hears dial tone, and the telephone set display clears. As the subscriber dials a DN, the digits are echoed on the lower display line. After the first digit is dialed, quiet tone replaces the dial tone heard by the subscriber. When dialing is complete, the subscriber hears call progress tones (such as audible ringback). If the call is answered, a voice path is established between the two parties. If the call is forwarded, the DN of the station to which the call is forwarded appears on the upper display line. To end a call, the subscriber either replaces the handset in its cradle or presses the *release* key on the business set.

When a call terminates to the PDN, the PDN LCD starts flashing. When a call terminates to an SDN, the associated SDN LCD starts flashing. After going off-hook, the called party presses the flashing key. If the incoming call is intra-group and the display set is idle, and if the set is not in a programming or digit collection state, the last four digits of the calling party's DN display on the upper display line. If the call is not intra-group and the display set is idle, a default message indicating information unavailable displays on the upper display line. In all other cases, the display doesn't change. To end a call, a subscriber either replaces the handset in its cradle or presses the *release* key on the business set.

When the CLASS on Centrex feature is activated in the switch, either four, seven, or ten digits display when the flashing key is pressed: four digits display for intragroup calls, seven digits display for DNs within the same area code, and ten digits display for DNs outside of the local area code.

### M5000-Series business set basic features

*On-hook dialing* enables the subscriber to originate calls without lifting the handset. To perform on-hook dialing, the subscriber presses an idle DN key. The associated DN LCD lights and the subscriber hears dial tone through the set speaker. Because the business set doesn't use DTMF signaling, the subscriber doesn't hear digit tones as the DN is dialed. When dialing is complete, the subscriber hears call progress tones (such as audible ringback) through the speaker. When the call is answered, the subscriber must converse with the called party using the handset; when the handset is lifted from its cradle, the speaker is disabled. To end the call, a calling party places the handset back in the cradle or presses the *release* key.

*Call Hold* enables a calling party to place an established call on hold for any length of time. After placing a call on hold, a calling party may then originate or receive another call on any other idle DN or perform other functions at the business set such as updating a Speed Call list. To place a call on hold, a calling party either presses the *hold* key, another DN key, or a feature key. The DN LCD associated with the held call starts flashing. To return to the held call, the calling party presses the flashing DN key. A single business set may have multiple calls on hold.

*Auto Hold* enables a DN on the set that is in talking state to be placed on hold automatically when the subscriber presses another ringing DN key during the call. Thus, simultaneous calls to a set can be answered more quickly and more easily.

*Listen on Hold* enables a calling party to place a call on hold, return the handset to its cradle, and listen through the speaker for the other party to return to the call. To activate the Listen on Hold feature, the subscriber presses the *hold* key; the DN key associated with the held call starts flashing. The subscriber then places the handset in its cradle and presses the flashing DN key. When the subscriber hears the other calling party's voice through the speaker, the call may be resumed using the handset.

The *Handsfree* feature enables a calling party to carry on a conversation through the microphone/speaker rather than the handset. To originate a Handsfree call, the subscriber presses the *handsfree* key and then dials the DN to be called. To end a call, the subscriber presses the *release* key or lifts the handset and then replaces it in its cradle.

The *Mute* feature enables a calling party to disable the microphone during a Handsfree call. *Mute* operation is applicable both to handsfree and regular calls. When the *mute* key is pressed, the subscriber's speaker or handset transmitter are disabled so that the party on the other end cannot hear the subscriber's voice. To deactivate the mute function, the subscriber presses the *mute* key again.

The *MBS Last Number Redial* feature enables the use of the ## digits to indicate that the last number dialed which resulted in a completed call should be redialed. Only valid DNs are stored; access codes or partially-dialed digits are not stored. Because this is a set-based feature, an M5000-Series set with multiple DNs still has only one DN stored as the LNR.

### M5000-Series business set add-on units

Three M5000-Series business set add-on units are available: M518 (18-key unit); M536 (36-key unit); M522 (22-key unit). The function of the add-on units is to increase the number of programmable MBS keys by a maximum of 52, and, in conjunction with the *Busy Lamp Field (BLF)* and *Direct Station Select (DSS)* features, to enable the M5000-Series business set to monitor stations within the same EBS group and provide attendant console features such as single-key operation for Call Transfer and Call Pickup.

### Busy Lamp Field (BLF)

The BLF feature enables an attendant to monitor the status of a station's DN: a designated DSS key lamp on the attendant's station flashes and the attendant hears a buzz tone when there is physical ringing at the monitored station; the lamp is turned on when the monitored station is busy on a call; the lamp is turned off when the monitored station is idle. To pick up a call at a monitored DN, the attendant goes off-hook or presses a DN key and then presses the flashing DSS key associated with the DN.

The following conditions pertain to BLF feature operation:

- The Automatic Callback, Automatic Recall, Call Park, Camp-on, and Ring Again features intentionally target a specific station. When a station is in ringing state due to one of these features, the associated DSS key lamp is not turned on. The lamp will be turned on, however, when the call is answered.
- A station can be monitored by up to 8 attendant stations. When the station rings, the associated DSS keys on the 8 attendant stations flash simultaneously.

- The DSS key lamp assigned to monitor a SIP station is not turned on when the SIP line goes off-hook. The DSS lamp is enabled when a call originated from a monitored SIP line rings at its destination.
- If a DSS-monitored SIP station calls a destination that is busy or otherwise unavailable, the DSS key lamp flashes momentarily but does not remain on for the duration of tone/announcement.
- If a DSS-monitored SIP station has activated the Do-Not-Disturb function through the SIP client device and a call is presented to the SIP line, the DSS lamp on the monitoring MBS set will receive brief alerting but cannot pick up the call.

### Direct Station Select (DSS)

The DSS feature enables the attendant to auto-dial monitored stations. To auto-dial an idle monitored station, the attendant goes off-hook or presses a DN key and then presses the DSS key associated with the monitored station. The monitored station is programmed by entering the 1 through 5-digit extension DN after the DSS mnemonic is entered as a key function. There can be a maximum of 2040 DSS keys per DMS-10 switch.

Using the auto-dial capability the attendant can also transfer a call to a monitored station, by pressing the *transfer* key and then pressing the appropriate DSS key.

### Add-on unit configuration

Add-on units can be used with only certain types of M5000-Series business sets. These restrictions are shown in Table 3-C.

**Table 3-C:**
**M5000-Series add-on unit compatibility**

| Add-on Module | M5000-Series set to which the add-on module can be assigned |
|---|---|
| M518 | M5009, M5112, M5209, M5312 |
| M522 | M5216, M5316 |
| M536 | M5009, M5112, M5209, M5312 |

An M5000-Series business line may be configured with a maximum of 63 keys. As shown in Table 3-D, add-on units can be daisy-chained together, using standard telephone cable, to provide a maximum of 52 add-on module keys.

**Table 3-D:**
**M5000-Series add-on units - maximum key assignments**

| Add-on Module | Maximum keys assignable |
|---|---|
| M518 | 52 (three M518 units connected together) |
| M522 | 44 (two M522 units connected together) |
| M518/M536 | 52 (one M518 unit connected together with one M536 unit) |

The unit numbers of the add-on units are automatically assigned in accordance with unit type and the order in which they are attached to an M5000-Series base set. The programmable key and unit number assignments for the three add-on unit types are shown in Table 3-E. M518 programmable key configurations are illustrated in Figures 3-7 through 3-9. M522 programmable key configurations are illustrated in Figures 3-10 and 3-11. M536 programmable key configurations are illustrated in Figures 3-7 and 3-12.

**Table 3-E:**
**M5000-Series add-on unit programmable key number and unit number**

| Unit Type | Feature Key Number | Unit Number |
|---|---|---|
| M518 | 12 through 29 | 1 |
| M518 | 30 through 47 | 2 |
| M518 | 48 through 63 | 3 |
| M522 | 16 through 37 | 1 |
| M522 | 38 through 59 | 2 |
| M536 | 30 through 63 | 1 |

**Figure 3-7: M518 programmable key configuration - first add-on unit**

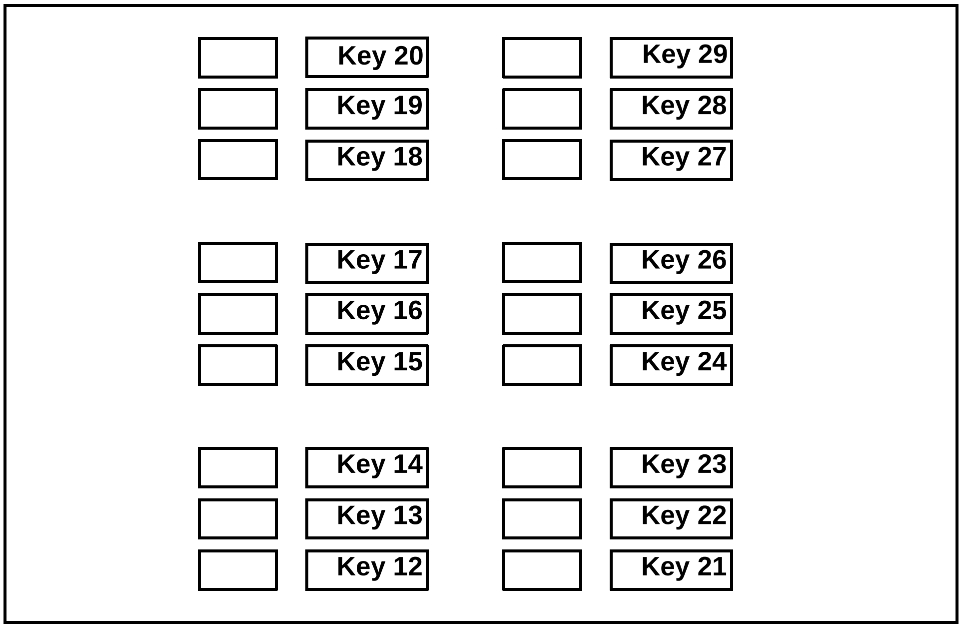



**Figure 3-8: M518 programmable key configuration - second add-on unit**

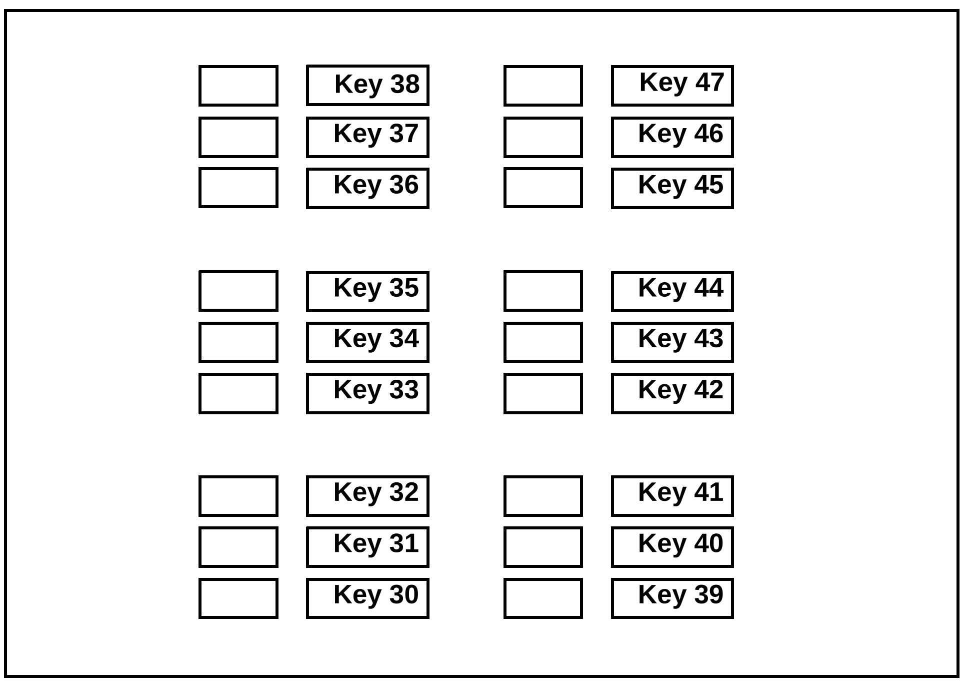


**Figure 3-9: M518 programmable key configuration - third add-on unit**

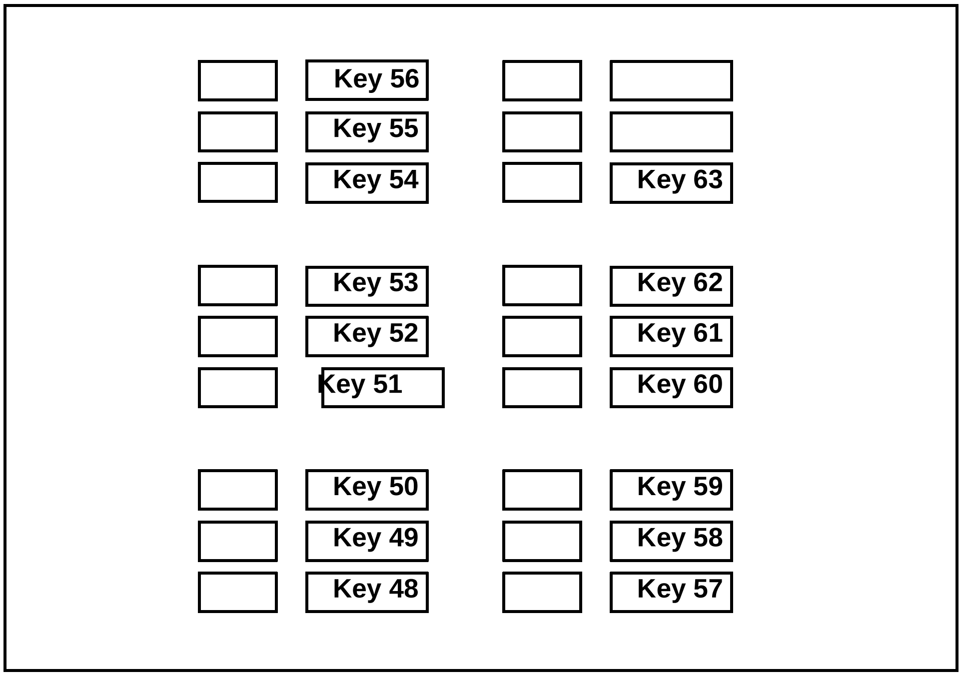



**Figure 3-10: M522 programmable key configuration - first add-on unit**

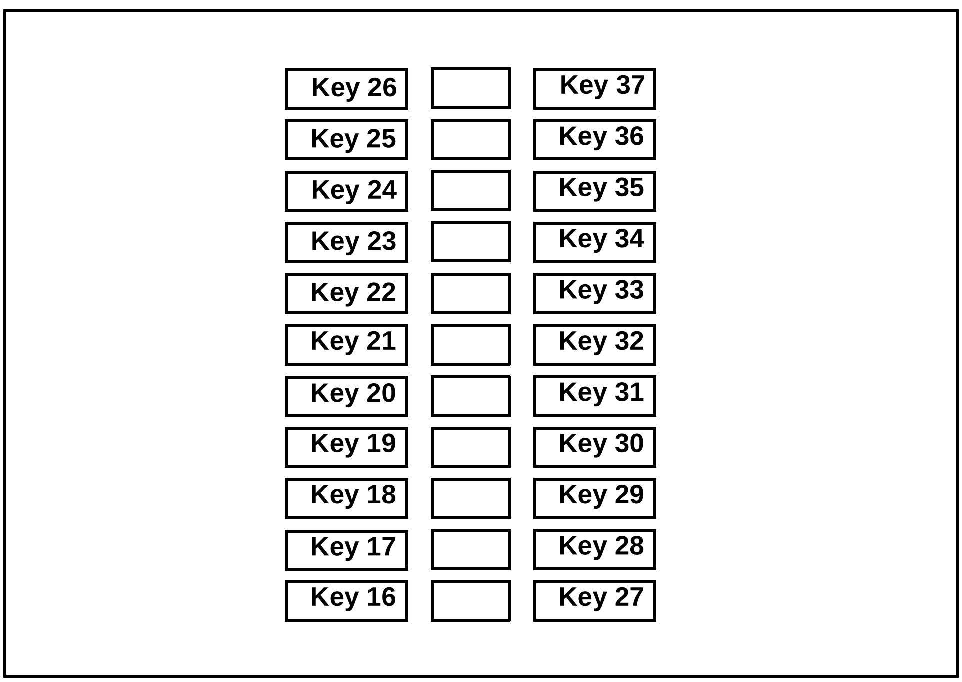


**Figure 3-11: M522 programmable key configuration - second add-on unit**

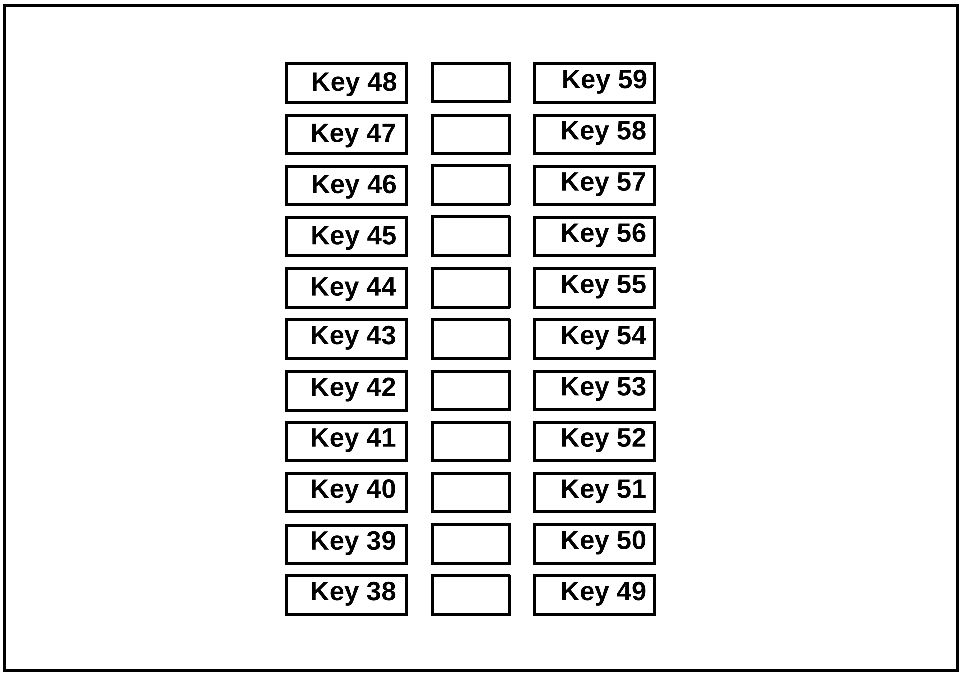
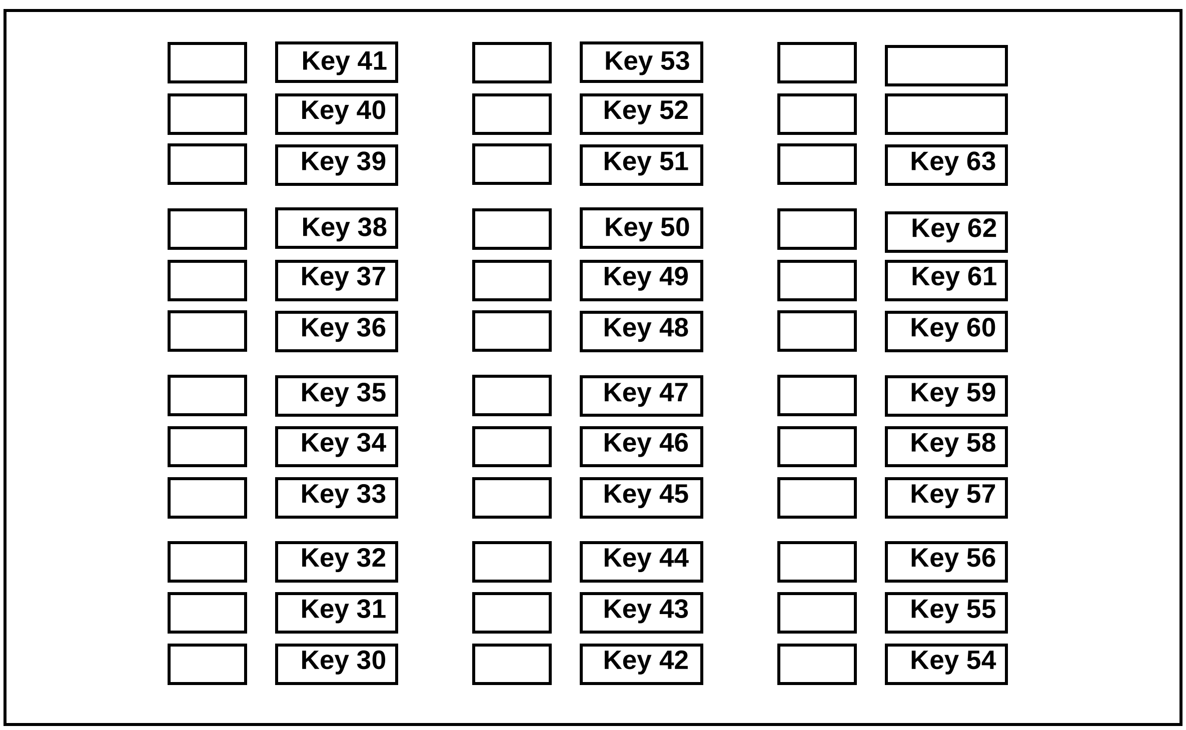



**Figure 3-12: M536 programmable key configuration - add-on unit**



## Automatic Dial (AUD)

The Automatic Dial feature enables an M5000-Series business set user to call a frequently dialed digit sequence by pressing an assigned AUD feature key. The digit sequence stored for the key, either a DN or an access code, is assigned by the subscriber or by operating company personnel.

### Programming the AUD key

When the M5000-Series business set is idle, the subscriber can press the AUD key to display the current digit stream assigned to it on the upper line of the telephone set display. If no digit stream is assigned, seven dashes display on the upper line of the telephone set display.

The subscriber may enter up to 32 digits (0 through 9, #, and *) either as a DN or as an access code. As the digits are entered, they display on the lower line of the telephone set display. After the 24th digit, the digits wrap around to the upper line of the display. To assign a digit sequence to the AUD key, the subscriber:

1) presses the AUD key to initiate the programming mode while the set is idle (all directory numbers and group intercoms on the set are idle or are on hold)

2) when the LCD associated with the key flashes, enters the number to be stored using the numeric keypad

3) presses the AUD key to cause the digits entered to be stored, after which the LCD turns off and the telephone set display clears

Pressing the Hold, RLS, I/C Group, or any DN key ends AUD key programming, and the digit stream just entered is not stored. Feature keys, including other AUD keys, are ignored during AUD key programming.

To delete a stored AUD key digit sequence, the subscriber:

1) presses the AUD key while the set is idle

2) when the AUD LCD begins flashing, enters #

3) presses the AUD key again, which completes the deletion and turns the LCD off

Unlike a conventional MBS, an ISDN terminal associates AUD to a combined Feature Activator/Feature Indicator (FAFI) assigned through Overlay ISDN, prompting sequence TCGN. The feature indicator provides visual indication and the feature activator provides the ISDN subscriber with the ability to have the digits dialed automatically. The FAFI position on a terminal is assigned to an ISDN line through the TCGN associated with a terminal service profile (TSP).

The subscriber may activate automatic dialing use the AUD FA under the following circumstances:

- dial tone state
- call forwarding access code special dial tone state
- in the initial call setup, for terminals that support call setup messaging

The subscriber can program the dialing sequence, while not in the context of a call, through the AUD non-call associated FA and entering up to 32 digits (0 through 9, #, and *). After entering the required digits, the AUD FA also ends the programming sequence. Stored AUD digits may be deleted by entering a single octothorpe (#) as the AUD dialing sequence. A dial access code is not supported for ISDN AUD.

### Using the AUD key

The subscriber may use the AUD key at any time during the following three call processing states:

- dial tone state (the subscriber goes off-hook, that is, presses a DN key)
- special dial tone state (the subscriber has activated a feature or entered an access code and has received special dial tone)
- established call state

If the subscriber presses the AUD key when no digits are stored against it, the key is ignored. If the subscriber presses a DN key, dials a partial number, and then presses the AUD key, a reorder tone is sent to the subscriber.

The AUD key may be used to dial feature access codes like Calling Number Delivery Blocking (CNB) or Calling Name Delivery Blocking (CNAB). An M5216 and M5316 business set feature enables combinations of authorization, or account, codes and DNs to be stored on a single key by using the *Pause* key.

On an ISDN terminal, if the subscriber activates Autodial from either a VI or CMD DNCT, then the number dialed should terminate to a DNCT of similar call type. For example, the subscriber would not want to autodial a data call type from a voice call type.

### CLASS on Centrex impact on AUD

The following paragraphs describe the interaction of Automatic Dial with individual Centrex features.

**Foreign Exchange Service (FX, FXA).** An FX subscriber may assign an FX access code to the AUD key, or an access code followed by a number to be dialed at the far CO end. Any digits assigned to the AUD key when that key is pressed after the FX facility is accessed are passed to the far CO end.

**Group Speed Calling (GSC), Individual Long List (LSC), Individual Short List (SSC).** Speed call activations may be assigned to an AUD key; the AUD key cannot be used, however, to update a number in a speed-calling list.

### CLASS feature interaction with AUD

**Automatic Recall (AR).** The second part of the two-stage AR activation cannot be performed from the AUD key.

**Customer Originated Trace (COT).** The second part of the two-stage COT activation cannot be performed from the AUD key.

**Screen List Editing (SLE).** The activation codes for the list commands cannot be performed from the AUD key.

## French/Spanish on Meridian Business Set

This feature enables all MBS sets within the same EBS group to receive message displays either in French, in English, or in Spanish. A mixture of display languages within the group is not permitted. The capability (see prompt LANG in Overlay HUNT (EBS)) is set up for the EBS group by the operating company, although English is the default language.

## Handsfree Auto Answerback (AAB)

The Handsfree Auto Answerback feature enables calls placed to a primary directory number (PDN) an idle M5000-Series business set with handsfree capability to be automatically answered. This feature can be useful in business groups where MBS subscribers find it physically difficult to either remove the handset or press the handsfree key on the MBS.

When a call is placed to a PDN on an MBS configured with AAB, the call is automatically answered after four seconds (two seconds of ringing followed by answer supervision two seconds after the ringing stops). When the call terminates, all PDN functions, such as displays, are provided. Conversation for the call takes place through a handsfree unit, although after the call has been answered, the handset can be picked up. When the call has remained in handsfree mode and the calling party hangs up, the call is automatically disconnected.

When a call is placed to a DN on the set other than the PDN, the call is handled normally. If the MBS is either active or in a state where a normal call would not terminate to it, AAB is not performed for a call placed to the PDN. Instead, normal termination treatment is provided.

AAB is activated by assigning the AAB option to a PDN. If an AAB key is then defined on the MBS, the subscriber can press the key to activate or deactivate the feature at any time, except during CFW, speed call (GSC, SSC, or LSC), or AUD feature programming requiring the use of MBS keys. When an AAB key is not assigned, AAB is always activated as long as the AAB option is assigned.

AAB can be configured only on MBS phones equipped with a handsfree unit (models M5112, M5312, M5316). AAB can be assigned only to Primary DN keys, but cannot be assigned to Multiple Appearance Directory Numbers (MADN). AAB can be provided to subscribers only on a flat-rate billing basis.

The following conditions pertain to the AAB feature:

- AAB key activation status is not saved as part of custom calling tape backup (CCTB) data; it will, however, be restored after a system reload (SYSLOAD) when the Auxiliary Data Store (ADS) is valid.
- AAB calls cannot be routed to voice mail.
- No billing records are created for AAB calls.
- The AAB feature requires NT6X21AC or later vintage line circuit packs.
- If any manual operations occur while AAB is active, incoming calls are not automatically answered.
- AAB does not override ringback for AR, ACB, PRK, DPRK, or CAMP.
- AAB overrides ESB ringback.
- AAB overrides CFW Don't Answer and DAT.
- AAB ringing overrides DSR and SDR.

## DMS-10 MDC Features available on M5000-Series business sets

The EBS features described below utilize the programmable keys of the M5000-Series business set.

### Call Forwarding (CFW)

The Call Forwarding (CFW) feature enables subscribers to forward calls to the PDN or SDNs of stations that have the CFW station option. Subscribers may activate Call Forwarding either by entering an access code followed by the DN to which calls are to be forwarded or by pressing the *Call Fwd* key, entering the forwarded-to DN, and then pressing the *Call Fwd* key again. To deactivate Call Forwarding, the subscriber either dials an access code and enters an octothorp (#) or presses the *Call Fwd* key, enters the octothorp, and then presses the *Call Fwd* key again. To reuse an existing number, the subscriber either enters the access code and the number used previously or presses the *Call Fwd* key and, if the correct forwarded-to number displays, presses the *Call Fwd* key again. When a call is forwarded, the originating party's station displays the DN originally called (on the bottom display line) and the DN to which the call was forwarded (on the top display line); the terminating party's station displays the DN of the originating party (on the top display line) and the first DN from which the call was forwarded (on the bottom display line).

### Call Park

Call Park enables members of an EBS group to park a call against their DN or another DN or GIC member number in the EBS group and continue to originate and receive calls. The Call Park feature is available in two versions: *Call Park* and *Directed Call Park*. *Call Park* enables subscribers to park a call placed to a member of their EBS group against their own DN or GIC member number while the called party is paged; *Directed Call Park* enables subscribers to park the call against the DN or GIC member number of the called party. For a description of Call Park and Directed Call Park operation using an M5000-Series set, see the feature description under the "Enhanced Business Services" heading in this section of this NTP.

### Call Pickup Group (CPUG)

The Call Pickup Group feature enables a subscriber to answer a call placed to an unattended station in the same call pickup group. To pick up a call, a subscriber presses a DN key and dials the call pickup access code. If more than one station in the call pickup group is ringing, the station that has been ringing the longest is picked up first. When a subscriber picks up a call, the originating party's display shows the DN of the station performing the call pickup and the DN of the station whose call was picked up; the display of the station performing the call pickup shows the DN of the originating station and the DN of the station whose call was picked up.

### Call Transfer Outside

As an enhancement to the Three-way Calling / User Transfer feature, Call Transfer Outside (CTO) enables a subscriber to transfer an established call to two subscribers outside of the EBS group. Feature operation is the same as that described for the Three-Way Calling / User Transfer feature.

### Call Waiting

The Call Waiting (CWT) feature alerts a subscriber who is already involved in a two-way call that another call is waiting. The feature also enables the subscriber to answer the waiting call while maintaining the existing call. The CWT feature comprises Call Waiting Incoming, Call Waiting Intragroup, Call Waiting Origination, Dial Call Waiting, and Cancel Call Waiting. *Call Waiting Incoming (CWTI)* alerts the subscriber when a third party from outside the EBS group is calling. *Call Waiting Intragroup (CWIG)* alerts the subscriber when a third party from inside the EBS group is calling. *Call Waiting Origination (CWTO)* imposes Call Waiting on another member of the same EBS group when that member doesn't have Call Waiting assigned. *Dial Call Waiting (DCWT)* performs the same function as CWTO except that an access code must be dialed to impose Call Waiting. *Cancel Call Waiting (CCWT)* enables a subscriber to prevent, on a per-call basis, any incoming or intragroup call from being given Call Waiting treatment. All of these features can be assigned to either the primary or secondary DN key. *Call Waiting Intragroup (CWIG)* can also be assigned to the *I/C Group* key.

A subscriber involved in a two-way call is informed of an incoming call either by ringback tone or special Call Waiting tone, or a blinking key. To answer the waiting call, the subscriber presses the *Call Wait* key. The subscriber may then return to the original call by pressing the key associated with the original call or by pressing the *Hold* key and then the key associated with the original party. The subscriber can also end the original call before picking up the waiting call.

### Camp-On

The Camp-On feature enables a member of an EBS group to use the Call Transfer feature to extend a call originating either from inside or outside of the group to a busy station. For a description of Camp-On operation using an M5000-Series set, see the feature description under the “Enhanced Business Services” heading in this section of this NTP.

### Group Intercom (GIC)

The Group Intercom (GIC) feature, which is optional for EBS group members, enables a subscriber to use abbreviated dialing to dial another member within the same EBS group. For a description of GIC operation using an M5000-Series set, see the feature description under the “Business Set Services” heading in this section of this NTP.

### Make Set Busy (MSB)

This feature enables subscribers to control the termination of calls on their Meridian Business Set. The Make Set Busy feature causes DN appearances, except MADNs, to be made busy to incoming calls. Two versions of the MSB feature are available:

- Make Set Busy All Calls (MSBA), which enables subscribers to make their MBS appear busy to all types of incoming calls, both external and internal
- Make Set Busy Intragroup (MSBI), which enables subscribers to make their MBS appear busy only to internal calls

The MSB features can be assigned only to MBS units that have EBS enabled.

While an MSB feature is active on a set, all incoming calls to DN appearances affected by the feature receive busy treatment. The set made-busy receives no indication that calls are attempting to terminate to it. Since MSBI involves only intragroup calls, calls received from outside of the group still terminate on the made-busy sets.

MSB can be activated or deactivated only by the use of a key, any time that a feature programming sequence is not occurring (for example, activating Call Forwarding or Speed Calling). Since MSB only affects call terminations, calls can still be originated from any of an MBS unit's DN appearances. Feature programming can also be performed on a made-busy set. If MSBA and MSBI are active at the same time, MSBA takes precedence over MSBI.

When MSB is active on an MBS, incoming calls still terminate to MADN members on the set. The MADN members do not, however, receive audible alerting. Instead, the visual indicator on the set associated with the MADN flashes, indicating that a call has terminated to the MADN. Since MSBI involves only intragroup calls, calls received from outside of the group still cause the group's sets to ring when they terminate.

The following conditions pertain to the MSB feature:

- Only one MSBA key can be assigned to an MBS.
- Only one MSBI key can be assigned to an MBS.
- MSB cannot be assigned to a 500/2500 telephone.
- Camp-On is not permitted, except for MADN members, on an MBS while MSB is active on the set.
- Call Waiting features are not permitted, except for MADN members, on an MBS while MSB is active on the set. If MSBI is active on the set, Call Waiting is not permitted for EBS intragroup calls to the set; external calls to the made-busy set may still use the Call Waiting feature.

### Message Waiting

Message Waiting enables subscribers to retrieve messages stored in the voice mail system. Message Waiting also notifies subscribers that messages are awaiting retrieval. For a description of Message Waiting operation using an M5000-Series set, see the feature description under the “Business Set Services” heading in this section of this NTP.

### Ring Again (RAG)

The Ring Again feature enables a subscriber who calls a busy station to be notified when the line is idle. To activate the RAG feature, the subscriber presses the *Ring Again* key. While the subscriber is waiting for notification, the business set may be used for other calls. When the busy station is free, the originating party receives an alerting tone and the LCD associated with the *Ring Again* key starts flashing. The subscriber then presses the appropriate DN key (*PDN*, *SDN*, *I/C Group*, *Conf 3*, or *Transfer*) and then presses the *Ring Again* key. If the subscriber is currently busy on all of the DN keys that could be used to perform the rering, generation of the alerting tone is postponed until a key is available.

### Speed Calling

Speed Calling enables a subscriber to maintain and use a speed calling list, which associates frequently called numbers with one- or two-digit index numbers. The Speed Calling capability comprises four features: Short Speed Calling, Long Speed Calling, Group Speed Calling, and Group Speed Calling Controller. *Short Speed Calling (SSC)* enables a subscriber to use a one-digit code for each of up to eight DNs. *Long Speed Calling (LSC)* enables a subscriber to use a two-digit code for each of up to 30 DNs. *Group Speed Calling (GSC)* enables the subscriber to use a two-digit code to access a speed calling list of up to 30 DNs in a single speed call group; *Group Speed Calling Controller (GSCC)* enables the subscriber to update this list.

To update a speed calling list, the subscriber presses the *Speed Call* key, enters the one- or two-digit code and the associated DN, and then presses the *Speed Call* key again. To place a call using a speed call list entry, the subscriber presses a DN key, then presses the *Speed Call* key, and then dials the speed call number.

For ISDN terminals, Speed Calling may be assigned using a Feature Activator (FA) and/or Feature Indicator (FI), or through a dial access code. An FA or FI must be used with a DN reference (FADN, FIDN, FFDN) assigned through Overlay ISDN, prompting sequence TCGN. Functionally the Speed Calling feature is identical to the conventional MBS speed calling feature. The feature indicator provides visual indication and the feature activator provides the ISDN subscriber with the ability to program speed calling numbers using a speed calling key.

### Three-way Calling / Consultation Hold

Three-way Calling (3WC) / Consultation Hold allows a subscriber to form a three-way call with two other parties located either within or outside of the EBS group. To add a third party to a call, the subscriber presses the *Conf 3* key. After receiving a special dial tone, the subscriber then dials the DN of the party to be added to the call; the subscriber's M5000-Series business set display shows the DN of the party on the lower display line. The subscriber may then speak privately with the third party (Consultation Hold feature) or may conference the call with the second party by pressing the *Conf 3* key. To alternate speaking with the second and third parties, the controlling party presses either the DN key associated with the second party or the *Conf 3* key (after the third party has answered). The controlling party must remain on line for the duration of the call to maintain the conference. To drop the third party from the call, the controlling party presses the *Conf 3* key.

### Three-way Calling / User Transfer

Three-way Calling / User Transfer enables a subscriber to include a third party in a three-way conference and then transfer the call to the third party. At least one of the two remaining parties involved with the call must be in the same EBS group as the party that transferred the call.

To transfer a call, the subscriber first presses the *Transfer* key and dials the DN to which the call is to be transferred. To conference the call with the second party, the subscriber presses the *Transfer* key again. To alternate speaking with the second and third parties, the subscriber presses either the DN key associated with the second party or the *Transfer* key (after the third party has answered). After the three parties have been included the call, the subscriber may either press the *RLS* key or hang up to complete the transfer of the call to the third party.

## CLASS on Centrex Features available on M5000-Series business sets

The CLASS on Centrex feature introduces Custom Local Area Signaling Services (CLASS) into the DMS-10 Meridian Digital Centrex (MDC) environment. The CLASS features described below utilize the special capabilities of the M5000-Series business set.

### Automatic Callback (ACB)

Automatic Callback enables a subscriber to redial automatically the last number called, whether the original call was answered, was unanswered, or encountered a busy signal. The ACB feature is assigned on a per-DN basis. To activate ACB, the subscriber presses the DN key assigned the ACB feature. The subscriber hears dial tone and the LCD associated with the key turns on. The subscriber then dials the activation service access code and the feature is activated toward the last DN called using the selected DN. If delayed processing is necessary, recall notification will occur when the called line and the DN assigned to the key are idle: the LCD on the DN key flashes and distinctive ringing is applied to the set (if ringing capability is not suppressed). If the subscriber has CND/CNAM, or if the original call was intra-group, the calling information identifying the DN being called displays on the telephone set display and, if the *reason display* option is enabled for the subscriber's group, the *reason* associated with the call displays below the calling information. When the subscriber presses the DN key, the called DN begins ringing.

### Anonymous Call Rejection (ACR)

Anonymous Call Rejection prevents anonymous calls from terminating at a subscriber's telephone set by routing these calls to a tone or announcement. To activate or deactivate the ACR feature on an M5000-Series telephone set, the subscriber presses a primary DN or secondary DN key assigned the ACR option and dials the appropriate ACR service access code.

### Automatic Recall (AR)

Automatic Recall enables a subscriber to return automatically the last incoming call, whether that call was answered or unanswered. The AR feature is assigned on a per-DN basis on an M5000-Series business set. Primary DNs and secondary DNs may be assigned either one-stage or two-stage feature activation. To activate AR, the subscriber presses a DN key to which the AR option is assigned. The subscriber hears dial tone and the LCD associated with the key lights. If the subscriber enters the activation service access code, the incoming call information associated with that MADN, or the DN, is retrieved and the call is placed. If delayed processing is necessary, only when the called line and the DN assigned to the key are idle does recall notification occur. At that time, the LCD associated with the DN flashes and distinctive ringing is applied to the set (if ringing capability is not suppressed). If the subscriber has CND/CNAM, or if the original call was intra-group, the information identifying the DN being called displays on the telephone set display and, if the *reason display* option is enabled for the subscriber's group, the *reason* associated with the call displays below the calling information. When the subscriber presses the DN key, the called DN begins ringing.

### Calling Number Delivery (CND)

Calling Number Delivery allows calling number information to be displayed at the called party's telephone set. CND and UCND are assigned on a per-DN basis on an M5000-Series business set. Although CND/UCND cannot be assigned directly to a GIC key, when the primary DN is assigned CND/UCND, the GIC key on the set will also have CND/UCND capability. When UCND is deactivated, normal MBS number display is provided for the DN to which UCND is assigned.

When CNAM is assigned to an MBS line, calling name information is displayed on the top line of the display following the directory number of the calling party. The *reason display* information (such as call forwarding information, call transfer information, or conference call information) is always displayed on the bottom line. Because only 24 characters can fit on a display line, the name portion of the display may be truncated depending on the length of the calling number. In order to reduce the name truncation, only 7 digits are displayed for calls that are from outside the Centrex group and are in the same HNPA as the group; the remaining characters are used for the display of the caller's name information. The public CND information will be either the station-to-station (STS) digits, for intra-group calls (calls within the same group dialed with either an STS or 9 + DN), 7/10 digits (for calls from outside the group), or the GIC number (for GIC calls). When the number is private, “PRIVATE#” displays; when the number is unavailable, “UNKNOWN#” displays. For both intra-group calls and calls from the network, the date and time of the call, which are usually provided for CND calls, are not displayed.

**Dialable Calling Number Delivery (DND)** Dialable Calling Number Delivery (DND) is an enhancement of the Calling Number Delivery (CND) feature. DND causes only the digits that must be dialed in order return the call to display rather than all ten digits normally displayed by CND. The DND display always takes precedence over the CND display.

DND can be assigned only to lines that are also assigned the CND/UCND station options.

### Calling Number Delivery Blocking (CNB)

Calling Number Delivery Blocking enables a subscriber to alter the privacy status of number information on a per-call basis. When CNB is activated, the permanent privacy status of the calling party is toggled for that call. CNB affects the number display for MBS only when CND is assigned. CNB does not override calling name and number display suppression for the Centrex group for intra-group calls (when prompt SNND = YES in prompting sequence EBS of Overlay HUNT). Dialing the CNB service access code is not allowed after a GIC call is initiated.

### Calling Name Delivery (CNAM)

Calling Name Delivery allows calling name information to be displayed on the called party's telephone set. CNAM and UNAM are assigned on a per-DN basis on an M5000-Series business set. Although CNAM/UNAM cannot be assigned directly to a GIC key, when the primary DN is assigned CNAM/UNAM, the GIC key on the set will also have CNAM/UNAM capability.

When CNAM is assigned to an MBS line, calling name information is displayed on the top line of the display following the directory number of the calling party. The *reason display* information is always displayed on the bottom line. Because only 24 characters can fit on a display line, the name portion of the display may be truncated depending on the length of the calling number. In order to reduce the name truncation, only 7 digits are displayed for calls that are from outside the Centrex group and are in the same HNPA as the group; the remaining characters are used for the display of the caller's name information. When the name is private, “PRIVATE NAME” displays; when the name is unavailable, “UNKNOWN NAME” displays. For both intra-group calls and calls from the network, the date and time of the call, which are usually provided for CNAM calls, are not displayed.

### Calling Name Delivery Blocking (CNAB)

Calling Name Delivery Blocking enables a subscriber to alter the privacy status of name information on a per-call basis. CNAB is assigned on a per-DN basis to DNs on an M5000-Series business set. When CNAB is activated, the permanent privacy status of the calling party is toggled for that call. CNAB does not override calling name and number display suppression for the Centrex group for intra-group calls (when prompt SNND = YES in prompting sequence EBS of Overlay HUNT). Dialing the CNAB service access code is not allowed after a GIC call has been initiated.

### Calling Identity Delivery and Suppression (CIDS)

When Calling Name and Number Delivery (CNND) or Calling Name and Number Blocking (CNNB) is activated, the permanent privacy status of the calling party name and number is "private/public" for that call. CIDS only affects MBS number display when CND is assigned. CIDS does not override calling name and number display suppression for the Centrex group for intra-group calls (when prompt SNND = YES in prompting sequence EBS of Overlay HUNT).

### Customer Originated Trace (COT)

Customer Originated Trace enables a subscriber to activate an immediate trace of the last incoming call. Primary DNs and secondary DNs may be assigned either one-stage or two-stage activation. To activate COT on an M5000-Series business set, the subscriber presses either a primary DN or secondary DN key assigned the COT option and then dials the COT service access code. The target DN of the trace is that of the last call placed to the primary DN.

### Screening List Editing (SLE)

Screening List Editing comprises four selective call services: Selective Call Acceptance (SCA), Selective Call Forwarding (SCF), Selective Call Rejection (SCR), and Selective Distinctive Ringing/Call Waiting (SDR). Each SLE feature is assigned on a per-DN basis on an M5000-Series business set. There is one screening list associated with each SLE feature assigned. When an incoming call arrives for a particular DN key, only the SLE lists associated with that DN key are examined.

Screening List Editing also includes the Simultaneous Ringing feature (SRNG). The SRNG feature may only be assigned to the Primary DN key on an MBS. There is a separate screening list associated with the SRNG feature for the Primary DN.

To add a DN to an SLE list, the may enter either a seven-digit DN, a 10-digit DN, an STS extension, if the number is prefixed with "02," or "01," to indicate that the DN of the last calling party (from the IMS) is to be added to the list.

When a call requiring Selective Distinctive Ringing treatment is placed to an M5000-Series business set that is idle, distinctive buzz tone is provided and the LCD associated with the DN flashes. If a call is in progress on the DN that is being called, distinctive call waiting tones, if configured, are applied to the line.

**Interaction between Screening List Editing and Call Waiting on an M5000-Series business set** Call Waiting treatment is not applied during an SLE editing session on a 500/2500 telephone; the calling party receives standard busy treatment.

Although normal Call Waiting treatment is applied during an SLE editing session on an M5000-Series business set, the session ends if the subscriber accesses the waiting call. If the subscriber requests that the last incoming call be added to the SLE list, the number of the waiting call, whether subsequently answered or not, will be added to the SLE list.

## Business Set Services

The Business Set Services software package, which enables the DMS-10 switch to support the Meridian Business Sets feature, includes features that utilize the special capabilities of the M5000-Series business set, described below.

### Group Intercom (GIC)

The Group Intercom (GIC) feature, which is optional for EBS group members, enables a subscriber to use abbreviated dialing to dial another member within the same EBS group. To have GIC capability, the subscriber must be a member of an intercom group within an EBS group. Up to twenty intercom groups, consisting either of up to 10 members with one-digit member numbers (0-9) or of up to 32 members with two-digit member numbers (00-31), may be associated with a single EBS group.

A GIC call can be placed from an M5000-Series business set using either on-hook dialing, handsfree dialing, or off-hook dialing. In order to place a GIC call, the subscriber presses the *I/C Group* key and dials the member number of a party in the same intercom group.

To place a GIC call from a 500/2500 telephone set, the subscriber goes off-hook and after receiving dial tone, dials the GIC feature access code followed by the member number of a party in the same intercom group.

For a procedure used to set up the GIC feature, see SOP 0136 in NTP 297-3601-311, *Data Modification Manual*

### Individual Page

Individual Page enables a subscriber to page another subscriber in the same intercom group using the speaker on an M5000-Series business set. To place a GIC call from the M5000-Series business set, the subscriber presses the *I/C Group* key and dials the member number of the party being called. After hearing ringback, the subscriber presses the *I/C Group* key again and then pages the called party. If the called party answers the page, a two-way speech path between the two parties is established.

### Message Waiting

Message Waiting enables subscribers to retrieve messages stored in the voice mail system. Message Waiting also notifies subscribers that messages are awaiting retrieval.

When a message is left in the voice mail system for any DN assigned to a given M5000-Series business set, the voice mail system sends notification to the DMS-10 switch over a Simplified Message Desk Interface (SMDI) link. For calls placed to any DN on the set except a secondary MADN DN, the DMS-10 switch then sends a message to the M5000-Series business set to turn on the *Msg Wait* LCD indicator. The subscriber at that set can then call the voice mail system and retrieve the stored voice mail. For additional information about the SMDI feature, see the feature description under the “Voice Mail” heading in this section of this NTP.

### Call Forward Reason Display (CFRD)

The Call Forward Reason Display (CFRD) feature enables a subscriber to display the reason for forwarding a call on the called party's display. When the subscriber forwards a call, the subscriber's display shows the DN originally called and the DN to which the call has been forwarded; the display of the party receiving the forwarded call shows the DN originally called and DN from which the call was forwarded followed by one of the following three reasons for the call being forwarded: “CALL FWD”, “BUSY FWD“, and “NO ANS FWD (no answer, forward).”

## Multiple Appearance Directory Number

The Multiple Appearance Directory Number (MADN) feature enables a DN to be assigned to a single EBS group consisting of up to eight members forming a MADN group. The MADN group may be configured with 500/2500-type sets, M5000-Series business sets, and with Voice over IP (VoIP) terminals in Generic 602.20 and later. Up to eight different MADNs can appear on a single M5000-Series business set. Each member of a MADN group may be assigned a different set of station options.

In Generic 602.20 and later, a MADN can also be assigned to an IBS group. The MADN group may be configured both with 500/2500-type sets and with VoIP terminals. The number of MADN groups was also increased to 16,384.

MADN groups are configured in a Single-Call Arrangement (SCA). In this configuration, only one M5000-Series business set may be active (either originating or terminating) on the MADN at a given time.

For a procedure used to set up the MADN feature, see SOP 0135 in NTP 297-3601-311, *Data Modification Manual*.

### MADN call processing

To originate a call from a MADN group, the entire MADN group must be call processing idle and the originating number must be in service. If the MADN is busy, the subscriber placing the call receives five seconds of reorder tone. If the MADN is not busy, the subscriber receives dial tone, indicating that the call can be placed; the DN key LCD indicator on all sets in that MADN group lights, indicating that the MADN is busy.

When a MADN is called, the DN key LCD indicator on all sets in that MADN group flashes. Ringing is started for members of the group who have the ringing option enabled. Any in-service member of the MADN group may answer the call. When the call is answered, ringing stops and the DN key LCD indicator on all sets in that MADN group lights. If the member answering the call is not the Primary MADN member, and the Primary MADN member is the Primary DN on a M5000-Series business set display, the display on the member's set clears.

### MADN Call Hold on M5000-Series business sets

A MADN group member who is originating a call may place the call on hold before an answer is received by pressing either the *Hold* key, a DN key, or the *I/C Group* key. On the member's M5000-Series business set, the DN key associated with the call being held begins flashing; the LCD indicator for the MADN on all other members' sets, however, remains ON. At this point, only the MADN group member who placed the call on hold may access the MADN. No indication is received by the originating party if the called party answers while the call is being held.

The MADN group member may also place a call that has been answered on hold by pressing either the *Hold* key, a DN key, or the *I/C Group* key. The DN key associated with the call being held begins flashing on all sets in that MADN group.

Any business set member of the MADN group can then access the MADN pressing the DN key associated with the held call.

### MADN Call Hold on 500/2500-type sets

A MADN group member may place a call that has been answered on hold by performing a switch-hook flash, dialing the “MADN hold” access code, and then going on-hook.

Any 500/2500 type member of the MADN group can then access the MADN by going off-hook.

### MADN Call Hold on Voice over IP (VoIP) terminals

For Voice over IP (VoIP) terminals, placing a call that has been answered on hold will depend on the VoIP terminal. For example, when a MADN group member is using a 2500-type set connected to a terminal adapter, the user would follow the step described above. When the VoIP terminal has multiple line or DN keys, A MADN group member would press the DN key, dial the “MADN hold” access code, and then going on-hook.

Any VoIP member of the MADN group can then access the MADN by going off-hook (dial tone is provided by the VoIP terminal) and dialing the “MADN hold” access code to retrieve the held call.

### Conditions applying to MADN VoIP terminals

The following conditions apply to the MADN feature:

- Because a proceed to dial indication such as dial tone is provided by the VoIP Session Initiation Protocol (SIP) device when the user goes off-hook or presses a line appearance key, the MADN group member must dial a “MADN hold” access code in order to retrieve a held call. The MADN hold and cancel hold access codes must be defined in translations. For details about the translations, refer to overlay TRNS in the NTP entitled *Data Modification Manual* (297-3601-311).

- When a SIP gateway line is assigned as the primary MADN group member and the Call Forward on Internet Down (CFID) feature is assigned and currently active, then calls terminating to the MADN DN will be forwarded when the primary MADN group member's SIP user agent is not registered; no MADN members will be alerted, including any wired lines. For more information see "Call Forward Internet Down (CFID)" on page 8-4.
- When a SIP gateway line is assigned as the primary MADN group member, then any features active on the user's gateway, such as call forwarding or do not disturb, take precedence over the alerting of the secondary MADN group members. Only when the gateway responds that it is alerting will the DMS-10 proceed to alert the other MADN group members.
- When a SIP gateway line is assigned as a secondary MADN group member, then any features active on the user's gateway, such as call forwarding or do not disturb, will cause that member to be passed by.

### Conditions applying to Calling Line Id on MADN terminals

Calling Line Id (CND - Calling Number Delivery and CNAM - Calling Name Delivery) may be assigned to the primary MADN group member with 500/2500-type sets, M5000-Series business sets, and with VoIP SIP terminals. Calling Line Id may also be assigned to the secondary MADN group members with M5000-Series business sets and with VoIP SIP terminals.

### Conditions applying to MADN

The following conditions apply to the MADN feature:

- When a call forwarding option (CFW, UCFW, CFB, UCFB, CFD, UCFD, CFF, UCFF) is moved to a “new” primary MADN as a result of a CHG or DEL MADN operation (see Overlay DN(MADN) in NTP 297-3601-311, *Data Modification Manual*), that option is deactivated and the ADS memory is released. The subscriber at the new primary MADN must activate the option.

### MADN in a Hunt Group

This feature allows MADN members to be included in DN hunt groups. When a call is placed to a hunt group containing MADN members, the busy/idle status of the MADN group determines whether the call can be presented to the group. If the call can be presented to the MADN group, the call terminates and is processed as a normal MADN group call; no additional hunting is performed. The Stop Line Hunt (SHU) and Random Line Hunt Make Busy (RMB) station options are assigned only to the Primary MADN member in the group and determine whether hunting will continue (SHU) or the line will be considered busy (RMB). If hunting cannot find an idle line/MADN group for termination, the pre-determined overflow treatment for the hunt group is performed.

- The Directory Number Hunt station options can only be assigned to the Primary MADN members in Overlay DN(MADN). Hunting order and overflow treatment are assigned to the hunt group through Overlay HUNT. More than one MADN group may be included in a single hunt group. A MADN may also be designated as the DNH overflow.

### Speed Calling on a MADN Group

Short Speed Calling (SSC), Long Speed Calling (LSC) and Group Speed Calling (GSC) can be assigned individually to primary or secondary MADN members. Either the primary or secondary MADN member can update his/her own list via a Screening List Editing (SLE) session. However, the SSC/LSC/GSC lists can not be updated via Web Control.

## CLASS on Centrex

The CLASS on Centrex feature introduces Custom Local Area Signaling Services (CLASS) into the DMS-10 Meridian Digital Centrex (MDC) environment. By offering CLASS features in both business and residential markets, an operating company can provide ubiquitous service to its entire body of CLASS subscribers, serving them both at work and at home from a single central office.

Essentially, CLASS on Centrex consists of a feature bit which, when enabled, allows the service provider to assign residential CLASS features already resident in the DMS-10 switch to the subscribers in one or more Centrex groups. Procedures for activation and deactivation of the CLASS features are the same as those used in the residential environment except that the *service access codes* are defined in the Centrex dialing plan. For service access codes that may be entered as a part of a call, the code must be entered before the directory number (for intra-group calls) or before the public network escape code (for calls exiting the group). Centrex subscribers are not allowed to activate/deactivate CLASS features using the public dialing plan.

In Generic 602.10 and earlier generic releases, CLASS on Centrex is available only to EBS groups. With the introduction of this feature, the number of EBS groups that can be served increases from 64 to 512. The total number of lines for all EBS groups assigned to a DMS-10 switch should not exceed 3000. In Generic 602.20 and later generic releases, CLASS on Centrex is also available to IBS groups.

CLASS on Centrex features are available both with 500/2500 telephone sets, Voice over IP (VoIP) terminals, and with M5000-Series business telephone sets (EBS groups only). Feature operation with 500/2500 telephone sets is described in the following paragraphs. Feature operation with M5000-Series business telephone sets is described in the section entitled, “CLASS on Centrex features available on M5000-Series business sets.” Feature operation with VoIP terminals will depend on the VoIP product and is not documented here as it is beyond the scope of this document.

## CLASS feature billing and configuration

All of the per-line CLASS features are available on a flat-rate and usage-sensitive basis. These features, which may be assigned only to non-coin, single-party lines, include: Anonymous Call Rejection (ACR), Automatic Callback (ACB), Automatic Recall (AR), Calling Number Delivery (CND), Calling Name Delivery (CNAM), Calling Number Delivery Blocking (CNB), Calling Name Delivery Blocking (CNAB), Calling Identity Delivery and Suppression (CIDS), Customer-Originated Trace (COT), Selective Call Acceptance (SCA), Selective Distinctive Ringing/Call Waiting (SDR), Selective Call Forwarding (SCF), and Selective Call Rejection (SCR).

ACR, CIDS, CNAB, CNB, and COT are available on an office-wide basis. The per-line denial features DACR, DCID, DCOT, and DNAB may be assigned to individual lines to prevent use of the office-wide capability from those lines.

Office-wide CNB (OCNB) and CIDS (OCID) are available on a flat-rate basis only. Office-wide ACR (OACR), CNAB (ONAB), and COT (OCOT) are provided on a usage-sensitive basis only. The per-line denial features are available on a flat-rate basis only.

Office-wide control of number delivery (prompts OSUP and TSUP in the SYS prompting sequence of Overlay CNFG) and name delivery (prompts ONAS and TNAS in the SYS prompting sequence of Overlay CNFG) are provided for calls entering and exiting the Centrex group from the public network and do not effect intra-group calls. Control of name and number delivery for intra-group calls is provided by a group-level parameter for suppression of number and name display (see prompts SNND and FNPR in Overlay HUNT (EBS) in NTP 297-3601-311, *Data Modification Manual*).

## Incoming/Outgoing memory slots

For M5000-Series business sets, the incoming memory slot (IMS), which contains calling display information, is configured on a per-set basis. Thus, primary DNs and secondary DNs assigned to one M5000-Series business set share the same IMS. When any DN on the M5000-Series business set requires IMS, all non-secondary MADNs on that set use the IMS. Each DN on an M5000-Series business set has an individual outgoing memory slot (OMS). A call incoming to the set on one of the set's DNs updates the IMS shared by all of the DNs assigned to the set. Calls from a DN on the set update only the OMS for that DN. Because the AR, COT, and SLE features use the IMS, the information from the last call made to the M5000-Series business set, rather than information from the last call made to a specific DN on the set, is used for feature operation. ACB, on the other hand, uses the OMS and information associated with the last call made from the DN, rather than information associated with the last call made from the set, for feature operation.

An IMS is configured on a per-MADN group basis; all group members share the IMS. Only the primary MADN (PMADN) is assigned the IMS; the secondary MADNs (SMADN) in the group use the IMS associated with the primary MADN. When IMS is required for the MADN group, the PMADN has IMS; any SMADNs that need IMS have access to the group's IMS. Each member of the MADN group has its own OMS; thus, outgoing calls from each member of the group update only the OMS associated with that member. Because the AR, COT, and SLE features use the IMS, when one of the features is activated from either a primary MADN or secondary MADN, the information from the last call made to the MADN group is used for feature operation. ACB, on the other hand, uses the OMS and information associated with the last call made from a particular MADN, rather than information associated with the last call made from the MADN group, for feature operation.

## Reason Display feature

The reason display feature is provided for the ACB and AR features on an office-wide basis. The operating company can either use the default display, “NOW AVAILABLE,” or create a display up to 24 characters long. The displays are used only during ACB and AR recall notifications to M5000-Series business sets in Centrex groups. Separate reason displays may be defined for each feature.

## Call Forward Privacy

The Call Forward Privacy feature is used to protect the privacy of the call originator when a call is forwarded. For example, if party A calls party B, who has activated call forwarding to party C, the DMS-10 switch may be configured to cause party A's name and number information to be unavailable (private) to party C. In this event, if party C has CNAM and/or CND assigned, the “private” indicator is displayed. The Call Forward Privacy feature applies to all forwarded calls in the office, whether residential or Centrex.

The following limitation applies to the call forward privacy feature:

- When a subscriber has activated call forwarding and call forward privacy is activated for the office, the call originator's name and number status is set to private. Since other features interact with the forwarded call as though the originator's name and number are private, some feature operation can be adversely affected. For example, if the forwarded-to subscriber has ACR activated, the forwarded call will be rejected and the call originator will be told to change their privacy status. Because call forward privacy is activated, the call originator will not be able to change privacy status for the call.

## CLASS on Centrex feature impact on MADN

The CLASS on Centrex feature introduces Custom Local Area Signaling Services (CLASS) into the DMS-10 Meridian Digital Centrex (MDC) environment. The following paragraphs describe the impact of the CLASS on Centrex features on the MADN feature.

**Anonymous Call Rejection (ACR)** ACR is assigned only to the primary member of a MADN group (PMADN) and only the primary member can activate or deactivate the ACR feature. All secondary MADN members use the ACR activation status of the PMADN. When ACR is activated by the PMADN member, the ACR feature is active for every other MADN member within the same MADN group. When ACR is deactivated by the PMADN member, then the ACR feature is deactivated for the whole MADN group.

**Automatic Callback (ACB)** The ACB feature is assigned on a per-MADN member basis. The outgoing call information is saved separately for each MADN member originating a call so that each member may used ACB independently to redial the last number dialed by that member. ACB feature activation and deactivation of pending requests affects only the MADN member using the service access code. Although up to 30 ACB requests may be pending for one MADN group, only one of the MADN members can be activating or deactivating, or participating in, an ACB call at one time.

If an ACB request is made by a MADN member when delayed processing is required, any subsequent attempts by other members of the MADN group to activate ACB against the same DN result in short-term denial treatment until the first member's request is complete.

Recall notification cannot occur until all members of the MADN are idle. Before recall notification occurs, the LCDs corresponding to that MADN on any other M5000-Series business sets are turned on to indicate that the MADN is in use. When recall notification is applied, the MADN member that originated the ACB request is alerted through special ringing.

**Automatic Recall (AR)** The AR feature is assigned on a per-MADN member basis. Some MADN group members may have one-stage activation capability while others may have two-stage activation capability. The target DN for an AR call is determined by the primary MADN member's last incoming call information (the last call that terminated to that MADN group). A MADN member may deactivate only those requests that the member initiates. Although up to 30 AR requests may be pending for one MADN group, only one of the MADN members can be activating or deactivating, or participating in, an AR call at one time.

If an AR request is made by a MADN member when delayed processing is required, any subsequent attempts by other members of the MADN group to activate AR against the same DN result in short-term denial treatment until the first member's request is complete.

Recall notification cannot occur until all members of the MADN are idle. Before recall notification occurs, the LCDs corresponding to that MADN on any other M5000-Series business sets are turned on to indicate that the MADN is in use. When recall notification is applied, the MADN member that originated the AR request is alerted through special ringing.

**Calling Identity Delivery and Suppression (CIDS)** CIDS is assigned on a per-MADN member basis for a MADN group. All of the MADN group members share the same station number and name as well as permanent number status and name privacy status.

**Calling Name Delivery (CNAM)** CNAM is assigned on an individual MADN member basis (to those members with sets capable of receiving the CLASS display information). When a call terminates to a MADN, calling name information is displayed at all MADN member locations where CNAM is activated. CNAM may not be assigned to a secondary MADN which is configured on a non-MBS telephone.

**Calling Name Delivery Blocking (CNAB)** CNAB is assigned on a per-MADN member basis for a MADN group. All of the MADN group members share the same station number and name as well as permanent number status and name privacy status.

**Calling Number Delivery (CND)** CND is assigned on an individual MADN member basis (to those members with sets capable of receiving the CLASS display information). When a call terminates to a MADN, calling number information is displayed at all MADN member locations where CND is activated. CND may not be assigned to a secondary MADN which is configured on a non-MBS telephone.

**Calling Number Delivery Blocking (CNB)** CBN is assigned on a per-MADN member basis for a MADN group. All of the MADN group members share the same station number and name as well as permanent number status and name privacy status.

**Customer Originated Trace (COT)** The COT/DCOT feature is assigned on a per-MADN member basis. Some MADN group members may have one-stage activation while others may have two-stage activation. COT can be performed from any MADN member station assigned the COT option. The target DN for the trace is the DN of the last call placed to the primary MADN.

The following limitation applies to COT when used in a MADN group. Since the incoming call information is altered with each incoming call and each call waiting notification, a MADN group with a high rate of incoming calls/call waiting notifications may be unable to invoke COT before the last call (malicious call) is updated. For a MADN group, the IMS associated with the primary MADN is used for the group. When the primary MADN is assigned to an M5000-Series business set, the IMS contains the information from the last call made to the set. Therefore, when COT is activated from a secondary MADN, the traced call may be one that did not terminate either to the MADN group or to the telephone used by the secondary MADN.

**Screening List Editing (SLE)** All SLE features except SRNG are assigned to MADNs on a primary member-only basis. The SRNG feature is not available to MADN members. All secondary MADNs are affected by the SLE feature use, however. Only the primary MADN member can edit the SLE lists, which apply to the entire MADN group. When adding or removing the DN of the most recent caller to a list, the DN is that of the most recent caller to the primary MADN. When a MADN is entered into an SLE list, all MADN group members are, in effect, added to the list.

Only the primary MADN group member receives reminder ring or buzz tone when a call has been received and forwarded through the Selective Call Forwarding (SCF) feature.

## CLASS on Centrex impact on ACR

For information about the impact of the CLASS on Centrex feature on ACR used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For the information about the interaction of ACR with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

The following paragraphs describe the interaction of Anonymous Call Rejection with individual Centrex features.

**Camp On** The presentation status of a subscriber using the Camp On feature is checked when the subscriber's call terminates to a subscriber with ACR active. For example, if subscriber A, who has the Camp On feature and a presentation status of “anonymous,” attempts to camp on to subscriber B, who has ACR active, the call is routed to rejection generic treatment. Otherwise, subscriber B can answer the call. If the camped-on call times out and the subscriber with Camp On is re-rung, the presentation status of the camped-on party is not checked.

**Foreign Exchange Service (FX)** If an FX subscriber wants to reject calls from the far CO end (the office hosting the subscriber's remote appearance), which may be marked private, the FXO station (remote appearance) must have the ACR feature active. Use of the ACR feature on the subscriber's local station end appearance will not reject anonymous calls from the far CO end since the incoming line-trunk call at the local station end (subscriber's local office) is considered a non-unique number. Similarly, any subscriber using ACR at the far CO end who receives a call from the FX subscriber will reject that call if the FXO line-trunk station is marked private, regardless of the privacy status of the FX subscriber's local station end appearance (that is, privacy indication is not passed across the FX facility).

**Inhibit Call Waiting (ICWT)** ACR takes precedence over ICWT. For example, if subscriber A, who has presentation restricted, imposes CWTO/DCWT on subscriber B, who has ACR active and is assigned ICWT, subscriber A's call is routed to rejection generic treatment. When the presentation status of subscriber A is changed to “public” and then subscriber A imposes CWTO/DCWT on subscriber B (with no change in the station configuration), subscriber A receives busy tone if subscriber B's station is not idle.

## CLASS on Centrex impact on ACB

For information about the impact of the CLASS on Centrex feature on ACB used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For the information about the interaction of ACB with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

### Interaction of ACB or Automatic Recall (AR) with Centrex features

The following paragraphs describe the interaction of ACB/AR with individual Centrex features.

**Busy Transfer (BTF), Busy Transfer All (BTFA), Busy Transfer Intragroup (BTFI)** An ACB/AR activation is not forwarded when it is attempted toward a subscriber with BTF, BTFA, or BTFI whose station is busy. For example, if subscriber A activates ACB/AR toward subscriber B, whose calls are being routed, through the BTF feature, to subscriber C, delayed processing occurs and subscriber A receives call notification when both parties are idle. If subscriber B also has ACB/AR and activates ACB/AR toward a busy station, the subsequent recall notification is applied to subscriber B only when subscriber B's station is idle; the ACB/AR recall notification does not follow the BTF call forward path to subscriber C.

**Don't Answer Transfer (DAT)** When subscriber A activates ACB/AR toward subscriber B, who has DAT activated toward subscriber C and whose station is idle, immediate processing occurs and the ACB/AR request is considered complete. After the selected number of ringing cycles, the call may then follow the call forward path. If subscriber B makes an ACB/AR request, recall notification is applied at subscriber B's station only; recall notification does not follow the call forward path to subscriber C even if subscriber B does not answer the recall notification.

**Call Hold (CHD)** If subscriber B, who has CHD, calls subscriber C and then puts subscriber C on hold, subscriber A's ACB/AR request against either subscriber B or subscriber C will receive delayed processing. If subscriber B disconnects, leaving subscriber C on hold, subscriber B receives re-ring treatment (by the CHD feature), and is, therefore, not idle until either subscriber C abandons the call or subscriber B answers the re-ring and then disconnects.

**Call Park** When subscriber C activates ACB/AR against subscriber A, whose call to subscriber B has been parked, subscriber C's request receives delayed processing. An ACB/AR request by subscriber C against subscriber B will receive immediate processing. If subscriber A's call is retrieved by subscriber D, subscriber A can later activate ACB/AR toward subscriber B. Subscriber B cannot activate ACB toward subscriber D. AR cannot be used to re-establish a call retrieved from being parked. Retrieving the parked call does not update subscriber D's IMS; therefore, if subscriber D retrieves the call from subscriber A, which has been parked by subscriber B, subscriber D cannot use AR to recall subscriber A.

**Call Pick-Up Group (CPUG)** ACB/AR recall notification to a subscriber is not permitted to be picked up by any CPUG subscriber even if the subscriber is part of the same call pick-up group. AR cannot be used to re-establish a picked-up call; if subscriber A calls subscriber B, and subscriber C, who is in subscriber B's call pick-up group, picks up the call, then subscriber C cannot use AR to re-call subscriber A. An ACB/AR call, however, may be picked up.

**Call Transfer (UTF); Call Transfer Outside (CTO)** If subscriber A calls subscriber B, who has UTF or CTO, and the call is then transferred to subscriber C, the IMS and OMS for the three subscribers are updated so that: 1) subscriber A can use ACB to call subscriber B (not subscriber C, to whom the call has been transferred); 2) subscriber B can use ACB to call subscriber C or AR to call subscriber A. Subscriber C can use AR to call subscriber B.

**Camp On** If subscriber A calls subscriber B, who camps the call onto subscriber C's station, which is busy, subscriber A can use ACB toward subscriber B (not subscriber C), subscriber B can use AR toward subscriber A and ACB toward subscriber C, and subscriber C can use AR toward subscriber B (not subscriber A). If subscriber A has the Camp On feature and has previously called subscriber B (who's station was busy) and was subsequently camped-on to subscriber B, subscriber A can activate ACB against subscriber B and subscriber B can activate AR against subscriber A.

**Directed Call Park (DPRK)** If subscriber A calls subscriber B, who parks the call with DPRK against subscriber D, subscriber C's ACB/AR request directed against subscriber A will receive delayed processing, but an ACB/AR request directed against subscribers B and C will receive immediate processing. When subscriber A's call is picked up by subscriber D, subscriber A can later activate ACB toward subscriber B (not subscriber D), subscriber B cannot activate ACB toward subscriber D but can activate AR against subscriber A, and subscriber D may not use ACB or AR to recall the retrieved party.

**Directed Call Pickup (DPUA), Directed Call Pickup With Barge-In (DCBI), Directed Call Pickup Without Barge-In (DCPU)** Subscriber C may pick up subscriber A's ACB/AR call to subscriber B, if subscribers B and C are in the same call pick-up group. ACB/AR recall notification to subscriber A is not permitted to be picked up by DPUA subscriber C, even if subscriber C is in the same call pick-up group as subscribers B and A.

**Directory Number Hunt (DNH)** Subscribers B, C, and D are members of a hunt group and have DNH. Line hunting proceeds from subscriber B's line, then to subscriber C's line, and then to subscriber D's line. If subscriber A activates ACB/AR toward subscriber B when only subscriber B's station is busy, delayed processing occurs and the request does not proceed to the next member of the group. If a hunt group member also has ACB/AR and initiates an ACB/AR request that requires delayed processing, busy/idle monitoring and recall notification are applied only to the specific hunt group member that initiated the request; no hunting occurs for application of the recall notification.

**Distinctive Ringing (DSR)** Distinctive ringing can be applied to a target station of an ACB/AR request if the station is assigned DSR. For example, if subscriber A activates an ACB/AR request toward subscriber B, who has DSR, and subscribers A and B are in the same Centrex group, subscriber B receives distinctive ringing. If subscriber B's station is busy and call waiting treatment is to be applied, then distinctive call waiting tones are given. Distinctive ringing or call waiting tones are not applied for recall notification.

**Foreign Exchange Service (FX, FXA, FXS, FXO)** If the FX subscriber does not use the access code method to reach the FX facility (FX option), activation of ACB/AR at the local station end is not possible. If the station is assigned the FXA option, which allows facility access by access code, ACB/AR may be activated on that subscriber's station. However, the FXA subscriber cannot use ACB or AR to re-call an FX call. ACB/AR are not compatible with the FXO option. ACB cannot be activated from an FXS line-type trunk station since this kind of station is used to connect FX subscribers to FX trunks.

**Group Intercom (GIC)** The only valid dialing sequences for GIC calls are member numbers within the same intercom group. Since ACB/AR require a dialed service access code for activation or deactivation, ACB/AR cannot be activated from the GIC key on an M5000-Series business set. The ACB/AR and GIC features are compatible on the same DN on a 500/2500 telephone set. However, because a GIC call does not update the IMS or OMS on either 500/2500 or M5000-Series business sets, an ACB/AR activation after a GIC call is made or received functions as though the GIC call has not occurred in that the recall is made on the last non-GIC call made to or from that set.

## CLASS on Centrex impact on AR

For information about the impact of the CLASS on Centrex feature on AR used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For information about the interaction of AR with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.” For a description of the interaction of AR with individual Centrex features, refer to the heading, “Interaction of ACB or Automatic Recall (AR) with Centrex features.”

The following limitations apply to AR operation in the Centrex environment:

- When SNND (Suppression of Name and Number Display) is activated, all intra-group calls are marked with *private* status. Thus, when the AR privacy feature is activated, AR cannot be used for intra-group calls.

## CLASS on Centrex feature impact on calling name delivery features

The following paragraphs describe the impact of CLASS on Centrex on calling name delivery group-level control and the interaction of CLASS on Centrex with individual calling name delivery features.

### Calling name delivery group-level control

Group-level control is provided for:

- the specific name that is delivered for intra-group calls when this name is stored in the DMS-10 switch
- the database to be used for intra-group calls, either local or centralized
- the name privacy for intra-group calls that are forwarded outside the group
- name display status for intra-group calls

Each Centrex group may subscribe to a Personal Name Delivery (PND) capability that applies to intra-group calls. With this capability enabled, an external database is consulted, through a TCAP query, for intra-group calls for that particular Centrex group. The TCAP query contains a business group identifier which is used to locate the correct name to be used for intra-group calls. The database to be used may be either *local* or *centralized*. Both types of database may be located at the same Service Control Point as the residential database. If the PND capability is not enabled, a single Centrex group name, assigned by the operating company and stored in the DMS-10 switch, is delivered for intra-group calls.

Control is provided for the per-call name privacy status delivered to a forwarded-to party through the *forced name privacy* (FNPR) control. When the control is enabled and one party of an intra-group call forwards the call outside the Centrex group, the per-call name privacy status delivered to the forwarded-to party is *private*. If the control is not enabled, the per-call name privacy status is not altered.

Each Centrex group may be configured to control the display of name and number information for intra-group calls. This group-level control supplements the office-wide ONAS and TNAS parameters (prompts ONAS and TNAS in the SYS prompting sequence of Overlay CNFG). ONAS and TNAS may be used to restrict the display of name information for non-intragroup calls, while the group-level control controls name information for intra-group calls. The group-level control is available to mark all intra-group calls as *private* for name and number or to allow normal intra-group display for name and number (see prompt SNND, Overlay HUNT (EBS), NTP 297-3601-311, *Data Modification Manual*).

### Interaction of CNAM or Calling Number Delivery (CND) with individual Centrex features

The following paragraphs describe the interaction of CNAM or Calling Number Delivery (CND) with individual Centrex features.

**Automatic Line (AUT), Denied Origination (DOR), and Manual Line (MAN)** There are no interactions between AUT, DOR, or MAN and flat-rate CND or CNAM. However, UCND and UNAM are incompatible with AUT, DOR, and MAN because the activation/deactivation codes for these features cannot be entered when AUT, DOR, or MAN is assigned to the line.

**Call Forwarding (CFW)** When subscriber A calls subscriber B, who has activated CFW to subscriber C, the following interactions exist when neither call forward privacy nor forced name privacy are activated:

- For 500/2500 telephone sets, subscriber B's display is unaffected by calls that are forwarded to subscriber C. Subscriber C's display is updated with subscriber A's number and/or name information (assuming subscriber C has CND and/or CNAM activated).
- For M5000-Series business sets, if subscriber C has CND and/or CNAM activated, subscriber A's CND/CNAM information is provided. In this case, CND/CNAM information for subscriber B is not provided.
- For 500/2500 and M5000-Series business sets, when call forward privacy is activated and subscriber C has CND and/or CNAM activated, subscriber A's name and/or number privacy status is set to “private” for display to subscriber C.
- The only additional interaction that occurs as the result of Busy Transfer (BTF), Busy Transfer All (BTFA), or Busy Transfer Intragroup (BTFI) is the indication of BTF activation in reason display output for subscriber C.

**Don't Answer Transfer (DAT)** When subscriber A calls subscriber B, who has activated DAT to subscriber C, the following interactions exist when neither call forward privacy nor forced name privacy are activated:

- For 500/2500 telephone sets, subscriber A's number and/or name information is displayed at subscriber B's station, if CND and/or CNAM are activated. If subscriber A's call is subsequently forwarded (when no answer occurs at subscriber B's station) to subscriber C, subscriber C's display is updated with subscriber A's number and/or name information (assuming subscriber C has CND and/or CNAM activated).

- For M5000-Series business sets, subscriber A's number and/or name information is displayed at subscriber B's station, if CND and/or CNAM are activated. If a 5000-Series business set does not have CND/CNAM display assigned when subscriber A's call is forwarded to subscriber C, subscriber A's display is updated with the number of the forwarded-to party and, when Reason Display is activated, “FORWARD” is also displayed. No CND or CNAM display information is provided to subscriber A. Subscriber C's display is updated with subscriber A's number, subscriber B's number and, when Reason Display is activated, “FWD NO ANS.” If subscriber C has CND and/or CNAM activated, subscriber A's CND/CNAM information is provided. In this case, CND/CNAM information for subscriber B is not provided.
- For 500/2500 and M5000-Series business sets, when call forward privacy is activated and subscriber C has CND and/or CNAM activated, the privacy indicator rather than subscriber A's display feature number and/or name information is provided to subscriber C. When Forced Name Privacy is activated and one party in an intra-group call forwards the call outside the Centrex group, subscriber A's CNAM privacy status is set to “private” for display to subscriber C.

**Call Hold (CHD)** If subscriber A calls subscriber B when subscriber B has subscriber C's call on hold, the following interactions exist:

- For 500/2500 telephone sets when subscriber B has CND and/or CNAM activated, subscriber A's calling number and/or name information is not displayed at subscriber B's station since subscriber B's station is not considered idle. If subscriber B goes on-hook while subscriber C's call is on hold, the recall ringing applied to subscriber B is not accompanied by subscriber C's calling number and/or name information (assuming subscriber B has CND and CNAM activated.) If subscriber A calls subscriber C when subscriber C's call is being held by subscriber B, subscriber C's station appears busy to the originator. Therefore, no CND/CNAM information is provided at subscriber C's display.

**Call Park** When subscriber C retrieves a call placed by subscriber A to subscriber B, who either parks the call (Call Park) or parks the call at subscriber C's station (Directed Call Park), the following interactions exist:

- For 500/2500 and M5000-Series business sets when subscriber B has CND and/or CNAM activated, no CND/CNAM information is displayed when subscriber C has CND and/or CNAM activated. When the parked call is recalled, if subscriber B has CND and/or CNAM activated, no CND/CNAM information is displayed.

**Call Pickup Group (CPUG), Directed Call Pickup Without Barge-In (DCPU), Directed Call Pickup Barge-In (DCBI), and Directed Call Pickup**

**From Any Station (DPUA)** When subscriber A calls subscriber B and subscriber C picks up the call, the following interactions exist:

- For 500/2500 telephone sets, subscriber B's display is updated with subscriber A's number and/or name information if CND and/or CNAM are activated on the station. When subscriber C picks up subscriber A's call to subscriber B, subscriber C does not receive subscriber A's calling number and/or name information (assuming subscriber C has CND and/or CNAM activated). If subscriber C barges into the call established between subscribers A and B, no special display interactions occur since transmission of calling information is not permitted when a set is off-hook.
- For M5000-Series business sets, if subscriber C has CND and/or CNAM activated, no CND/CNAM information is displayed.

**User Transfer (UTF) and Call Transfer Outside (CTO)** When subscriber A calls subscriber B, who transfers the call to subscriber C, the following interactions exist:

- For 500/2500 telephone sets, subscriber B's display is updated with subscriber A's number and/or name information if CND and/or CNAM are activated on the station. When subscriber B transfers the call to subscriber C's station, subscriber C's display is updated with subscriber B's number and/or name information if CND and/or CNAM are assigned to the station.
- For M5000-Series business sets not assigned CND/CNAM display, subscriber B's display is updated with subscriber A's number. If subscriber B has CND and/or CNAM activated, subscriber A's CND/CNAM information is displayed. For M5000-Series business sets without CND/CNAM display assigned, when subscriber B transfers the call to subscriber C, subscriber C's display is updated with subscriber B's number. If subscriber C has CND and/or CNAM activated, CND/CNAM information for subscriber B is displayed.
- For M5000-Series business sets without CND/CNAM display assigned, before subscriber B drops out of the call, subscriber B may toggle between subscribers A and C. Subscriber B's display is updated with the previous display provided for the active party (subscriber A or subscriber C).
- For M5000-Series business sets without CND/CNAM display assigned, when subscriber B drops out of the call, subscriber A's display is updated with subscriber C's number and, when Reason Display is activated, “CALL TRANSFER” is also displayed. No CND or CNAM display information is provided to subscriber A. When subscriber B drops out of the call, subscriber C's display gets updated with subscriber A's number and, when Reason Display is activated, “CALL TRANSFER” is displayed. If subscriber C has CND and CNAM activated, no CND and/or CNAM information is provided for subscriber A.

**Call Waiting (CWT), Call Waiting Incoming (CWI), and Call Waiting Intragroup (CWIG)** When subscriber B calls subscriber A, who has a call waiting feature assigned and is in talking state, subscriber A is alerted about the waiting call either by a call waiting tone, for 500/2500 telephone sets, or by a flashing CWT LCD and call waiting tone on an M5000-Series business set. If subscriber A has a 500/2500 telephone set, the subscriber's display is not updated because the subscriber is off-hook. If subscriber A goes on-hook while subscriber B is waiting, subscriber B's number and/or name information is not displayed when re-ring occurs. If subscriber A has an M5000-Series business set, subscriber A's display is updated with subscriber B's number when subscriber A answers the call (M5000-Series business sets without CND/CNAM display assigned) or, if subscriber A has CND and/or CNAM activated, subscriber A's display is updated with subscriber B's CND/CNAM information.

**Camp On** When subscriber C retrieves a call placed by subscriber A to subscriber B, who subsequently camps a call onto subscriber C's station, the following interactions exist:

- For 500/2500 and M5000-Series business sets, when subscriber C has CND/CNAM activated and retrieves the call, no CND and/or CNAM information is displayed. When the camped-on call is recalled, and if subscriber B has CND and/or CNAM activated, no CND/CNAM information is displayed.

**Cancel Call Waiting (CCWT)** There are no interactions between CCWT and flat-rate CND or CNAM. A subscriber with CCWT and UCND or UNAM is not permitted to dial the CCWT service access code followed by the UCND or UNAM activation code. If the subscriber attempts this, standard dialing error treatment is applied to the call.

**Denied Termination (DTM)** DTM is compatible with the display features; however, DTM prevents termination to the station.

**Directory Number Hunting (DNH)** When subscriber A calls a member of a hunt group, normal CLASS display set updating occurs when subscriber A is connected with one of the hunt group members. Subscriber A's CND/CNAM information is displayed when CND and/or CNAM are activated on the terminating line. When subscriber A calls from an M5000-Series business set, subscriber A's display is not updated.

**Disable Ringing (DSRG)** Delivery of calling party information to terminals that have CLASS display capability occurs between the first and second ring for 500/2500 telephone sets. If a DN does not have ringing capability (for example, option DSRG is assigned to a MADN member), the calling party CND/CNAM information is not delivered to that member. Ringing has no effect on display of CND/CNAM information on an M5000-Series business set.

**Foreign Exchange Service (FX, FXA, FXS, FXO)** Calling party information is not passed across the FX facility. An FX subscriber with CND and/or CNAM receiving a call from the far CO end receives an unknown caller indication as normally supplied with trunk calls, not the far CO end calling party information. The display features are not compatible with the FXS or FXO line-type trunk stations since no equipment is attached to these lines to receive display information. Nor can the information be passed across the FX facility to the subscriber for incoming FX calls since the display features require the delivery of inband Frequency Shift Keying (FSK) tones to a terminator during the first silent period of a ringing phase. This is not possible over a device that is signaled as a trunk.

**Group Intercom (GIC)** To provide display features for a GIC, the CND and/or CNAM station options are assigned to the DN on 500/2500 telephone sets, or to the PDN on M5000-Series business sets. The following are specific interactions between GIC and CND, and between GIC and CNAM:

- For CND, the GIC number is displayed in the same location as that used for non-GIC CND display on the specific telephone type. When a GIC call terminates on a 500/2500 telephone set, the DN of the call originator (if the caller is using a 500/2500 telephone set), or the PDN of the call originator (if the caller is using an M5000-Series business set) is displayed instead of the GIC number. For intra-group calls terminating at a 500/2500 telephone set, the complete DN is displayed instead of STS digits.
- For CNAM, the name associated with the DN or with the PDN is displayed in the same location as that used for non-GIC CNAM display on the specific telephone type.

**Group Speed Calling (GSC), Long List Speed Calling (LSC), and Short List Speed Calling (SSC)** When the GSC, LSC, or SSC features are used to terminate a call to a line on which CND and/or CNAM are activated, the calling information that is displayed is the same as that which would be displayed if the call was dialed without using the GSC, LSC, or SSC features. UCND or UNAM activation/deactivation codes may be included in the GSC speed calling list.

**M5000-Series business sets** When subscriber A calls subscriber B and subscriber B has subscriber C's call on hold, the following interactions exist:

- Subscriber B's display is updated according to the usual incoming call interactions. After subscriber B answers the call from subscriber A, subscriber B may toggle between subscribers A and C. For M5000-Series business sets without CND/CNAM display assigned, subscriber B's display is updated with the previous display provided for the active party (subscriber A or C). If subscriber A puts subscriber B on hold and calls subscriber D, subscriber A may toggle between subscribers B and D. Subscriber A's display is updated with either the original calling digits (subscriber B's number) or subscriber D's number; no CND/CNAM display information is provided to subscriber A.

**Multiple Appearance Directory Number (MADN)** When a call terminates to a MADN, CND/CNAM information is displayed at all MADN member locations where CND and/or CNAM are activated. When the call is put on hold and another MADN member retrieves that call, the member's terminating display is that normally provided when the member retrieves a call.

**Ring Again (RAG)** CND/CNAM information can be displayed upon recall notification for Centrex subscribers using 500/2500 sets. CND/CNAM information is not provided upon recall notification for Centrex subscribers using M5000-Series business sets.

**Three-Way Calling (3WC)** When subscriber A calls subscriber B, subscriber B calls subscriber C, and then subscriber B creates a conference call among subscribers A, B, and C, the following interactions exist:

- For 500/2500 telephone sets, subscriber B's display is updated with subscriber A's CND/CNAM information if CND and/or CNAM are activated on the station. When subscriber B calls subscriber C, subscriber C's display is updated with subscriber B's CND/CNAM information if CND and/or CNAM are activated on the station. When subscriber B creates a conference call among the three subscribers, there is no change in the displays for the three subscribers.
- For M5000-Series business sets, when subscriber B creates a conference call among the three subscribers, “CONFERENCE” displays on the top line of subscriber A's display and on the top line of subscriber B's display. This display overwrites any existing CND/CNAM information displayed on the sets. If subscriber C has already received subscriber B's CND/CNAM information, “CONFERENCE” displays on subscriber C's set.

**Voice Mail (MD, MWIL, SIDT, SMDI)** CNAM may not be assigned to an SMDI line because that kind of link is not equipped to accept name information. CND, however, may be assigned to an SMDI line. There are no interactions between MD, MWIL, SIDT and the display features.

**Warm Line (WARM)** There are no interactions between WARM and flat-rate CND or CNAM. WARM provides a delay of from 2 through 30 seconds before the call is routed; therefore, the UCND or UNAM activation/deactivation codes may be entered from a line assigned WARM.

## CLASS on Centrex impact on CND

Each Centrex group may be configured to control the display of name and number information for intra-group calls. This group-level control supplements the office-wide OSUP and TSUP parameters (prompts OSUP and TSUP in the SYS prompting sequence of Overlay CNFG). OSUP and TSUP may be used to restrict the display of number information for non-intragroup calls, while the group-level control controls name information for intra-group calls. The group-level control is available to mark all intra-group calls as *private* for name and number or to allow normal intra-group display for name and number.

For information about the impact of the CLASS on Centrex feature on CND used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For information about the interaction of CND with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

For a description of the interaction of CND with individual Centrex features, refer to the heading, “Interaction of CNAM or Calling Number Delivery (CND) with individual Centrex features.”

## CLASS on Centrex impact on CNB

For information about the impact of the CLASS on Centrex feature on CNB used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For information about the interaction of CNB with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

### Calling display blocking/delivery codes (CNB, CNAB, CIDS) interactions with Centrex features

Calling display blocking/delivery codes may be entered in any compatible order and are normally entered before other service access codes. Any exceptions to the order in which the service access code is entered is noted in the following paragraphs.

**Automatic Line (AUT)** Calling display blocking/delivery codes are incompatible with AUT.

**Call Forwarding (CFW), Busy Transfer (BTF), Busy Transfer All (BTFA), Busy Transfer Intragroup (BTFI), and Don't Answer Transfer (DAT)** If subscriber A dials a calling display blocking/delivery code prior to calling subscriber B, who has activated a call forwarding feature to subscriber C, then subscriber A's name and/or number information is not toggled if either call forward privacy or FNPR is activated. CNB, CNAB, or CIDS will not override call forward privacy or FNPR.

**Call Hold (CHD)** Dialing a calling display blocking/delivery code prior to dialing the service access code results in a dialing error.

**Call Park (PRK)** Dialing a calling display blocking/delivery code prior to dialing the service access code or pressing an MBS key results in a dialing error.

**Call Pick-Up Group (CPUG)** Dialing a calling display blocking/delivery code prior to dialing the service access code or pressing an MBS key results in a dialing error.

**User Transfer (UTF)** A subscriber may dial a calling display blocking/delivery code prior to attempting to transfer the call to a third party.

**Call Transfer Outside (CTO)** A subscriber may dial a calling display blocking/delivery code prior to attempting to transfer the call to a third party outside the Centrex group.

**Camp On** Dialing a calling display blocking/delivery code prior to dialing the service access code or pressing an MBS key results in a dialing error.

**Cancel Call Waiting (CCWT)** A subscriber may dial a calling display blocking/delivery code prior to dialing the service access code.

**Denied Origination (DOR)** Calling display blocking/delivery codes are incompatible with the DOR feature.

**Dial Call Waiting (DCWT)** A subscriber may dial a calling display blocking/delivery code prior to dialing the service access code.

**Directed Call Park (DPRK)** Dialing a calling display blocking/delivery code prior to dialing the service access code or pressing an MBS key results in a dialing error.

**Directed Call Pick-Up Barge-In (DCBI)** Dialing a calling display blocking/delivery code prior to dialing the service access code results in a dialing error.

**Directed Call Pick-Up Without Barge-In (DCPU)** Dialing a calling display blocking/delivery code prior to dialing the service access code results in a dialing error.

**Group Intercom (GIC)** When a subscriber using a 500/2500 telephone set dials a calling blocking/delivery code prior to dialing the service access code, a dialing error results. If a subscriber dials a calling blocking/delivery code from the GIC key on an M5000-Series business set, a dialing error results.

**Group Speed Calling (GSC), Long List Speed Calling (LSC), Short List Speed Calling (SSC)** A subscriber may dial a calling display blocking/delivery code prior to initiating a speed call. Display blocking/delivery codes may be included in the speed call list.

**Manual Line (MAN)** Calling display blocking/delivery codes are incompatible with the MAN feature.

**Multiple Appearance Directory Number (MADN)** Calling display/delivery codes are assigned to MADN members on a per-MADN member basis. A MADN member may dial a calling display blocking/delivery code prior to dialing the called party's number.

**Ring Again (RAG)** On a 500/2500 telephone set, a subscriber may dial a calling display/delivery code prior to dialing the service access code. The calling display blocking/delivery code will override any calling display blocking/delivery code(s) that were dialed on the original call. If a subscriber uses the RAG key at an M5000-Series business set, a calling display blocking/delivery code cannot be entered.

**Three-Way Calling (3WC)** A subscriber may dial a calling display/delivery code prior to attempting to add a third party to the call.

**Toll Restricted Services (TDN)** A subscriber may dial a calling display blocking/ delivery code prior to initiating a call on a line with the TDN option.

**Warm Line (WARM)** A subscriber may dial a calling display blocking/delivery code prior to initiating a call on a line with the WARM option because a WARM line provides a delay of from 2 through 30 seconds before the call is routed. Calling display blocking/delivery codes may be programmed in with the WARM line digits.

### Calling display blocking/delivery codes (CNB, CNAB, CIDS) interactions with CLASS features

*Anonymous Call Rejection (ACR).* The ACR feature rejects an incoming call if, by activation of CNB, CNAB, or CIDS, the incoming number/name are considered anonymous.

**Automatic Callback (ACB)** A subscriber may dial a calling display blocking/ delivery code prior to dialing the ACB service access code. The code entered affects the presentation status of the current ACB request and overrides any calling display blocking/delivery code(s) dialed during the original call or ACB activation.

**Automatic Recall (AR)** A subscriber may dial a calling display blocking/delivery code prior to dialing the AR service access code. The code entered affects the presentation status of the current AR request and overrides any calling display blocking/delivery code(s) dialed during the original call or AR activation.

**Calling Display Features (CND, CNAM)** A subscriber may dial a calling display blocking/delivery code prior to dialing the called number to alter the calling subscriber's presentation status. At an M5000-Series business set, dialing a calling display blocking/delivery code prior to dialing the called number has no effect on the calling subscriber's presentation status if the called party does not have CND and/or CNAM on the line.

**Customer-Originated Trace (COT)** Dialing a calling display blocking/delivery code prior to dialing the service access code results in a dialing error. If a subscriber dials a calling display blocking/delivery code to make the presentation status private for a particular call, COT can still be used by the called party to trace the calling party's call.

## CLASS on Centrex impact on COT

For information about the impact of the CLASS on Centrex feature on COT used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For information about the interaction of COT with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

### Interaction of COT with individual Centrex features

The following paragraphs describe the interaction of COT with individual Centrex features.

**Call Park and Directed Call Park (DPRK)** When subscriber A calls an attendant who has Call Park and COT, and wants to extend the call to subscriber B, the attendant has the ability to activate COT against subscriber A. After the call is established between subscribers A and B, subscriber B may not activate COT against subscriber A.

**Call Pickup Group (CPUG)** When subscriber C picks up subscriber A's call to subscriber B, subscriber C does not have the ability to activate COT against subscriber A. Subscriber B may not activate COT against subscriber C. Subscriber B may, however, activate COT against subscriber A.

**M5000-Series business sets** If COT is assigned to an M5000-Series business set DN, the COT subscriber may put a call on hold by pressing the set's *hold* key and selecting another DN to dial the service access code to activate COT.

**Multiple Appearance Directory Number (MADN)** COT is assigned on a per-MADN member basis. COT can be performed from any MADN member who has COT. This is achieved by retrieving the last incoming call information from the incoming memory associated with the primary member of the MADN group.

## CLASS on Centrex impact on SLE

For information about the impact of the CLASS on Centrex feature on SLE used in conjunction with the Meridian Business Sets feature, refer to the heading “CLASS on Centrex features available on M5000-Series business sets.” For information about the interaction of SLE with the Multiple Appearance Directory Number (MADN) feature, refer to the heading, “CLASS on Centrex feature impact on MADN.”

### Interaction of all SLE features with individual Centrex features

The following paragraphs describe the interaction of all SLE features with individual Centrex features. Except where noted the Simultaneous Ringing Feature is not included in the SLE feature interactions.

**Call Park** The parked caller's DN is not checked against the SLE lists of the party retrieving the parked call. The DN of the party retrieving the parked call is not checked against any SLE lists of the parked caller.

**Call Pickup Group (CPUG)** The DN of a party using CPUG to pick up an incoming call is not checked against the SLE list of the calling party.

**User Transfer (UTF)** On 500/2500 telephone sets and M5000-Series business sets, if a subscriber who is assigned the UTF option transfers a call to a subscriber who is assigned an SLE feature, the DN of the subscriber transferring the call is checked against the SLE list of the subscriber to whom the call is transferred. The DN of the subscriber whose call is being transferred is not checked. In addition, a subscriber with both SLE and UTF is not allowed to transfer a call to a party involved in an SLE list editing session for all SLE features including SRNG, nor can a subscriber transfer an editing session to another subscriber.

**Camp On** If a subscriber camps on a call to a subscriber assigned an SLE feature, the DN of the party camping on the call is checked against the SLE list of the subscriber. The DN of the party whose call is being camped on is not checked.

**Directed Call Pickup Without Barge-In (DCPU)** If a subscriber with SLE and DCPU attempts to pick up a call for another line, the DN of the incoming caller is not checked against the subscriber's SLE list.

**Directory Number Hunting (DNH)** DNs of incoming calls to a hunt group are checked against the SLE list of the subscriber whose DN is being called. The DN of an incoming call is checked only against the SCA and SCR lists of subscribers in the group whose DNs are subsequently hunted for the call.

**Group Intercom (GIC)** SLE feature station options may not be assigned to a GIC key on an M5000-Series business set. A GIC number cannot be added to an SLE list. Although SLE can be assigned to 500/2500 sets assigned the GIC feature, the SLE feature lists associated with the sets are not checked for GIC calls.

**Three-Way Calling (3WC)** A subscriber with SLE features including SRNG and 3WC may perform SLE feature list editing while talking to another party by dialing the SLE service access code on the second leg of the call.

### Interaction of Selective Distinctive Ringing/Call Waiting (SDR) with individual Centrex features

The following paragraphs describe the interaction of SDR with individual Centrex features.

**Disable Ringing** Disable Ringing takes precedence over SDR when features are active on a MADN. Distinctive notification is not applied to MADN members with Disable Ringing.

**Distinctive Ringing (DSR)** SDR takes precedence over DSR. If a non-intragroup call is received, distinctive notification is given to the called party. If the call is an intragroup call, but the caller's DN is on the called party's SDR screening list, distinctive call waiting tones are applied to the station when that station is busy.

## Web Based Feature Control

The Generic 505.10 Web Based Feature Control (WEBF) allows subscribers to maintain their CLASS on Centrex, Simultaneous Ringing, and Speed Call lists via a third party interface. List update are made through a web page that the service provider maintains. The service provider enables some or all of the features assigned to the subscriber on the DMS-10 for update via the Web page.

The DMS-10 will interface to the subscriber's web interface through a intermediate SCP/Adjunct. The DMS-10 interfaces to the SCP with Common Channel Signalling (CCS) links. The Service Node (SN) will send the DMS-10 instructions to update the subscriber lists using TCAP Packages as defined in Telcordia GR-1299 AINGR: Switch - Service Control Point (SCP)/Adjunct Interface.

Specifically, a subscriber would be able to make the following changes to their CLASS on Centrex and Simring lists:

- Activate the feature
- Deactivate the feature
- Add a number to the list
- Add the last calling party to the list
- Add a speed calling entry to the list
- Delete a number from the list
- Delete all private entries from the list
- Delete all entries from the list
- Query the list entries

- Query the list size
- Query feature activation status

Additionally for Selective Call Forwarding the user can:

- Set the redirection number
- Set the redirection number to a speed call entry
- Clear the redirection number
- Update the redirection number

For Long Speed Calling, Short Speed Calling, Group Speed Calling, and Convenience Dialing the subscriber can:

- Set an index
- Query their speed call list

The WEBF feature will have not impact on the billing. The DMS-10 will not track messages sent to the DMS-10 for billing purposes.

## Centrex IP

Centrex IP is a Voice over Internet Protocol (VoIP) telephony service that serves small and large businesses and the mobile work force. With the increase in data network usage, small and large businesses require high-performance data access technology for all their communications needs. Centrex IP integrates seamlessly with existing corporate networks to unify the delivery of voice and data over IP connections. It allows voice and data traffic to travel over a variety of carrier grade, cost-efficient packet networks. With Centrex IP, service providers can offer the same feature-rich capabilities of Centrex over an IP network, and still have the quality, security, and reliability of the public switched telephone network (PSTN). Business travelers and telecommuters can access the same business services they have in the office from any location where they have broadband access.

Centrex IP coexists within current Centrex groups and share the same dialing plans, line features, group features, and network access. This allows service providers to retain and grow their existing Centrex markets.

Benefits to service providers are listed as follows:

- Leverages DMS-10 investment by extending DMS-10 Centrex onto the growing IP market
- Allows smooth migration of DMS-10 customers to managed IP networks

- Provides seamless integration with existing IP networks, eliminating network duplication
- Provides migration strategy to next generation data services

Centrex IP offers the following benefits to end users:

- Provides same features and benefits as DMS-10 Centrex
- Available to both Integrated Business Services (IBS) and Enhanced Business Services (EBS) customer groups
- Allows one network for voice and data, reducing network administration and expense
- Provides the platform for advanced services deployment while utilizing existing infrastructure
- Extends business communication services to the mobile work force
- Permits end user terminals to be moved without service provider assistance
- Allows a mixture of VoIP terminals, 2500 sets, and Meridian Business Sets within a customer group

For information on Centrex group and station features that are available to VoIP subscribers, see "Centrex IP" on page 100.

# Section 4: Line features

## Miscellaneous line features

For more detailed information on the implementation of the line features introduced in this section, see the NTP entitled *Data Modification Manual* (297-3601-311). For a description of each type of line available with a DMS-10 switch, refer to NTP 297-3601-150, *Equipment Identification*. For details about ANI and ringing codes, refer to NTP 297-3601-180, *System Performance Specifications*. For detailed line-interface information, see NTP 297-3601-184, *Circuit Interfaces for Lines, Trunks, and Test Trunks*.

### Automatic Number Identification (ANI)

ANI is a means of identifying the calling station for billing purposes. The number to be billed is automatically forwarded to Automatic Message Accounting (AMA) equipment. This feature is available only on single-party, two-party, and four-party lines.

### Automatic Number Identification Failure (ANIF)

ANI failure occurs when automatic number identification (ANI) has failed to identify the calling number of a customer-dialed toll call. The call is routed through the route assigned to the ANI failure generic condition. Up to four ANI failure routes may be assigned, if the ANI Failure Route Confliction feature is configured.

#### Automatic Number Identification Failure Route Confliction (ARC)

Prior to the ANI Failure Route Confliction feature, all ANI failure (ANIF) traffic was routed through one route, and all *no CAMA position available* (NCPS) traffic was routed through one route. ARC allows one ANIF route and one NCPS route to be defined for each home numbering plan area (HNPA) that is configured in the DMS-10 switch.

Each ANIF and NCPS route is associated with an HNPA. Up to four ANIF routes (ANIF, ANF2, ANF3, and ANF4) and up to four NCPS routes (NCPS, NCP2, NCP3, and NCP4) may be defined. During call processing, the route that has been defined for the call originator's HNPA will be used to route the ANIF or NCPS call.

## Flexible Automatic Number Identification (FANI)

FANI is an enhancement of the Automatic Number Identification (ANI) feature. FANI enables the telco to define the identity digit, or ANI ID code, portion of the ANI dialing pattern. The two-digit ANI ID code provides the InterLATA or International Carrier (IC) with information about the a-party.

With FANI, the telco is able to define up to 100 ANI ID codes which may be assigned to residential or Integrated Business Services (IBS) stations and outgoing Enhanced Business Services (EBS) Virtual Facilities Groups (VFG). Operator Services (OS), Traffic Services Positioning System (TSPS), Local Equal Access System (LEAS), and Equal Access (EQA) routes may be designated by the telco to carry FANI codes. If, however, an IC chooses not to receive FANI codes, the telco is able to prevent the codes from being sent over the designated routes. The FANI ID codes do not replace any of the existing hard-coded ANI ID codes. FANI codes do, however, override the hard-coded ANI ID codes.

### FANI for incoming trunk groups

An enhancement to the FANI feature enables FANI codes to be assigned to incoming or two-way trunk groups. FANI can be assigned only to trunk groups that are supported by the Line Featured Trunk patch.

## Circle Digit Translation (CDIG)

The CDIG feature is a means of identifying a calling station on a multiparty line for billing purposes. A unique party identification digit (circle digit), 0 through 9, is assigned to each party for up to 10-party service. After accessing the Direct Distance Dialing (DDD) network, the calling party dials the circle digit, then the called number. The DMS-10 switch produces the billing number automatically and forwards it to AMA equipment.

## Extended Range Lines (ERL)

The ERL feature allows extension of the supervision range on the subscriber loop to 4500 (including the telephone set). This extension is achieved by switching the -48 V and ground to -48 V and +48 V, respectively, when the loop is greater than 1900 (including the telephone set).

## Foreign Exchange Service

The system can provide service to a customer located outside the normal serving area through a combination of switched access through the DMS-10 switch and a dedicated transmission facility.

## Private branch exchange (PBX)

In the DMS-10 system, a PBX trunk is usually served by a Miscellaneous Line circuit pack. The trunk may be equipped for either loop or ground start. Battery reversal may be provided to cause message register operation or toll diversion.

PBX features available with this configuration are: Line Hunting (Random-Make-Busy and Stop-Hunt), Direct Inward Dialing, Toll Restriction (Toll-Denied and Toll-Diversion), hotel/motel message registers, hotel/motel routing to a TSPS, and overflow registers.

### Second Dial Tone

Second Dial Tone allows dial tone to be returned to a calling line after certain digits have been dialed by the subscriber.

## Station options

This subsection briefly describes some of the subscriber station options that can be applied to subscriber lines, either by the customer or through the DMS-10 switch. A complete listing is presented in Overlay DN, prompting sequence STN under the OPT prompt.

### Automatic Line (AUT)

When a call is originated on an automatic line, the call is routed to a specified terminating number, either within the office or, by way of an outgoing trunk, to another office. A terminating call is processed normally unless the line is programmed *denied-terminating*.

### Coin, Coin First (CCF)

Dial tone is provided to the coin telephone set when the initial rate has been deposited. The coins provide a circuit to ground to operate the ground-start line. Coin collection and return are controlled by the DMS-10 switch in the following manner:

- The coins are collected at the end of a non-operator call. (Coins can be collected or returned at the end of dialing an operator call.)
- If no connection can be established, the coins are returned after the handset has been placed on hook.
- If the call is to a free number, the coins can be returned at the end of dialing or at the end of the call.
- If the call is an INWATS (1+800+ 7 digits) call, the coins are returned and an access billing record is generated.

### Coin, Dial Tone First (CDF)

Dial-Tone-First coin service uses a loop-start line to provide dial tone with no prior coin deposit. After a sufficient number of digits has been dialed to determine whether or not the call is directed to a free number, the central office makes a ground test for initial deposit. If the initial rate has been satisfied, or if the call is directed to a free number, the connection is completed. The coins are returned or collected at the end of the call or at the end of dialing, depending on the type of call.

## Coin, Semipostpay (CSP)

Dial tone is provided to the coin telephone set when the handset is lifted. When a call is dialed and the called party answers, a line reversal is applied to the coin telephone set to block voice transmission until the initial rate has been deposited. On calls to free numbers, no reversal signal is returned to the coin station. All deposited coins proceed directly to the coin box and cannot be returned to the caller.

> ***Note:*** *The CSP station option is not available on SLC-96 coin lines.*

## Customer Assignable Options (OPTN)

This feature allows operating companies to have up to four station options that they can assign their own meaning to. These options are then used by the operating company to provide certain stations with custom routing of calls while also allowing more flexible translations for operating company needs. The station options assigned by the operating company are compatible with all other station options offered by a particular generic, and can be assigned in the same way as other station options. See the NTP entitled *Data Modification Manual* (297-3601-311) for more details about the procedure that is used to assign these and other station options.

## Customer Assignable Station Options, enhanced (CASO)

This feature increases the number of customer-assignable station options (see the description above) by 64. Each of the additional 64 station options may be named by the operating company.

## Denied Originating (DOR)

A line with the DOR feature cannot originate a call. An off-hook routes all attempted originations to a generic route as defined in office data. This feature cannot be applied to multiparty lines.

## Denied Terminating (DTM)

Lines with the DTM feature cannot receive a terminating call. All incoming calls are routed to a generic route as defined in office data.

## Digitone (DGT)

A subscriber's line circuit may be programmed to accept Digitone signals from the telephone set. Dial pulse and Digitone stations may both share the same line, as long as it is programmed for Digitone.

## Directory Number Hunting (DNH)

The DNH feature permits a search for an idle directory number (DN) within a group of DNs (called a *Hunt Group*) in order to complete an incoming call. The DMS-10 switch can provide this feature for 256 stations in each of DNH groups. 2047 DNH hunt groups are available. Any one of the following four types of directory number hunting can be provided to a Hunt Group:

- first hunt: hunting begins at the first DN in the list.
- sequential hunt: hunting begins at the dialed DN and continues to the end of the list.
- circular hunt: hunting begins at the dialed DN and continues once through every DN in the list.
- distributed hunt: hunting begins at the next DN idle during the last termination and continues once around the list.

CFGDA (Call Forward Group Don't Answer) is a feature that is provided through a new prompt in the OVLY HUNT DNH prompting sequence. The new prompt is FGDA. FGDA can be set to forward the call to the next DN in the hunt group (i.e. hunt on busy or idle DN) or forward the call to the pilot's voice mail when an idle member is found. CFGDA is only activated when the pilot is first called in the hunt group.

When a busy, non-pilot DN is called directly and CFB is enabled, hunting will not occur. Instead, the call is forwarded to the DN programmed.

### Distinctive Ringing on Single Party Revertive calls (DRR)

This feature provides the distinctive ringing pattern (0.5 seconds on - 1 second off - 0.5 seconds on - 4 seconds off) for revertive calls on all extensions of a single-party line. The feature can be assigned either to all stations in an office or to individual stations.

### Free Number Terminating (FNT)

Local coin or message-rate calls completed to a station with this feature are free of charge.

### INWATS (IWT)

Measured INWATS service is only available if the DMS-10 switch is equipped with LAMA. Toll-band screening for INWATS is not provided by the DMS-10 switch. Local calls to an INWATS line may optionally be blocked or allowed.

### Line Hunting, Random Make-Busy (RMB)

The RMB feature allows a predetermined group of lines to be busied by operating a key located at a PBX. The list of lines to be busied is modifiable through DMO. This feature affects only terminating calls; the lines may still be used for outgoing calls. A Miscellaneous Line pack (NT2T44) is required for this feature for PE, a Type B Line card (NT6X18) is required for this feature for LCE, and an S203 Line pack is required for this feature for SLE. For a procedure to be used for setting up the RMB feature see Service Order Procedure (SOP) 0016 in NTP 297-3601-311, *Data Modification Manual*.

## Line Hunting, Stop Hunt (SHU)

The SHU feature is activated by using a key located at a PBX. When activated, line hunting stops at a line that has been preselected and defined in data. A Miscellaneous Line pack (NT2T44) is required for this feature for Peripheral Equipment (PE), a Type B Line card (NT6X18) is required for this feature for Line Concentrating Equipment (LCE), and an S203 Line pack is required for this feature for Subscriber Loop Equipment (SLE).

## Manual Line (MAN)

When a call is originated on a manual line, the call is routed to the operator. A terminating call is processed normally.

## Operator number identification (ONI)

When the calling number cannot be identified automatically, the calling number is identified by an operator.

## OUTWATS (OWTF and OWTM)

Full business day (OWTF) and measured time (OWTM) OUTWATS service is routed through AMA equipment, where the appropriate billing information is recorded. An OUTWATS line is denied incoming calls. All calls to local or Extended Area Service (EAS) calling areas, to INWATS numbers, or to WATS bands not purchased by the subscriber are routed as defined in office data.

## Permanent Signal (PSIG)

When one party remains on-hook after the other party has disconnected from a call, the PSIG station option specifies that the party's station is be provided with permanent signal generic condition rather than dial tone.

## Station Suspended Options (SUS, SUSO, SUST)

The Station Suspended Options feature is an enhancement to suspended service and is used to suspend stations from origination only (SUSO), termination only (SUST), or both origination and termination (SUS). A new prefix translator may be provided to allow limited dialing to stations suspended from origination by way of SUSO or SUS. All other calls are given generic-route treatment. A “restore” request removes all suspension information from the station. All station options assigned to a station remain unchanged by the suspension and restoral process.

The following conditions apply to Station Suspended Options:

- A station that does not have Automatic Number Identification (ANI) cannot be set with origination suspension.
- A Remote Call Forwarding directory number cannot be set with origination suspension.

- A station with Call Forwarding activated and origination suspension may not cancel the Call Forwarding unless the origination suspension prefix translator is designed to handle a deactivate Call Forward access code.
- Stations with origination suspension and limited dialing may not access features that are facilitated in the IBS, EBS or Custom Calling prefix translators unless the origination suspension prefix translator is designed to handle the feature.
- EBS stations with origination suspension and limited dialing may not be able to dial emergency numbers unless the origination suspension prefix translator is designed to handle DOD digits.

## Teen Service (TEEN)

This feature allows a station to have two directory numbers on the same single-party line so that a subscriber can receive calls dialed to separate numbers without installing a second line. The subscriber is issued a *primary directory number* (PDN) and a *secondary directory number* (SDN). Calls placed to the PDN are identified by normal ringing, while calls to the SDN are identified by distinctive ringing. Although calls can terminate to either an PDN or to an SDN, they can originate only from the PDN. All billing is applied to the PDN.

### Enhanced Teen Service

The Enhanced Teen Service feature enables a single station line to be assigned up to four TEEN DNs. Calls to any one of the four TEEN lines will ring with a cadence unique to that particular TEEN DN, as shown in Tables 4-A and 4-B. Calls placed to a busy TEEN DN that has a call waiting option will cause a call waiting tone unique to that particular TEEN DN to be applied, as shown in Table 4-C.

**Table 4-A:**
**Enhanced Teen Service teen ringing patterns for ring types, by line equipment type**

| Line Equipment Type | Coded RInging | MFR Ringing | MFBE Ringing | Superimposed Ringing |
|---|---|---|---|---|
| RSC-S | short - long | short - short | short - long | short - long |
| LCM | short - long | short - short | short - long | short - long |
| SLC | long - long | short - short | short - short | short - short |
| RCU | long - long | short - short | short - short | short - short |
| IDT | long - long | long - long | long - long | long - long |

| Table 4-B:<br>Enhanced Teen Service coded ringing patterns for TEEN, TN2, TN3, and TN4 DNs, by line equipment type | | | | |
|---|---|---|---|---|
| **Line Equipment Type** | **TEEN Coded RInging** | **TN2 Coded Ringing** | **TN3 Coded Ringing** | **TN4 Coded Ringing** |
| RSC-S | short - long | short - short - long | short - long - short | short - short |
| LCM | short - long | short - short - long | short - long - short | short - short |
| SLC | long - long | short - short - long | short - long - short | short - short |
| RCU | long - long | short - short - long | short - long - short | short - short |
| IDT | long - long | short - short - long | short - long - short | short - short |

| Table 4-C:<br>Enhanced Teen Service Call Waiting tone cadences by DN type | |
|---|---|
| **TEEN DN Type** | **Tone Pattern** |
| TEEN | short - short |
| TN2 | short - short - short |
| TN3 | short - long - short |
| TN4 | long - long |

Enhanced Teen Service is available only to LCM, RSC-S, IDT, RCU, and SLC lines in offices configured for coded ringing. The TEEN DNs have the same options as the PDN.

The Enhanced Teen Service feature also provides the telco the capability of suppressing call waiting from individual TEEN DNs associated with a primary DN that has the CWT option. When the Call Waiting Autosuppression (CWAS) option is enabled for a TEEN DN, calls to that TEEN DN will not be interrupted by call waiting tones if a call is placed to the line while the line is involved in a call; calling parties attempting to call that line receive busy tone instead of ringback tone.

**Enhanced Teen Service Call Forwarding and Voice Mail** If the PDN has a call forwarding option, calls placed to the station's TEEN DNs can be forwarded to the number specified for the PDN. The Enhanced Teen Service feature enables each TEEN DN (including TEEN, TN2, TN3, TN4) to be assigned a voice mailbox. The following call forwarding options are available for each TEEN DN:

- calls to the TEEN DN are never to be forwarded
- calls to the TEEN DN should be forwarded, and, if also forwarded to a VMS, any messages left by the caller are placed in the PDN's voice mailbox

- calls to the TEEN DN should be forwarded, and, if also forwarded to a VMS, any messages left by the caller are placed in the TEEN DN's voice mailbox
- calls to the TEEN DN should be forwarded only if the call gets forwarded to a VMS; any messages left by the caller are placed in PDN's voice mailbox
- calls to the TEEN DN should be forwarded only if the call gets forwarded to a VMS; any messages left by the caller are placed in the TEEN DN's voice mailbox

### Conditions pertaining to Teen Service

The following conditions apply to Teen Service:

- The features, Cancel Call Waiting, Speed Calling, and Three-way Calling, are available for lines assigned the Teen Service feature.
- Teen Service is not compatible with Meridian Digital Centrex lines.
- Teen Service can be applied only to single-party lines.
- If coded ringing is configured, RNG, TIP, R1, or T1 are the only ring codes that may be assigned to lines with TEEN or TN4. If MF ringing is configured, RNG or TIP are the only ring codes that may be assigned. If SIMP ringing is configured, RNG, TIP, R1, R2, T1, or T2 are the only ring codes that may be assigned.
- Options TN2 and TN3 are not compatible with the RCO (ring codes) option on NT6X17 lines.

## Toll Denied (TDN)

The TDN feature restricts a line from originating dialed toll calls. Any attempted toll-call originations are intercepted by the DMS-10 switch and routed to an announcement or tone (as defined in office data).

## Warm Line (WARM)

When a call is originated on a warm line (that is, the subscriber goes off-hook), the subscriber is given dial tone and then the call is automatically routed to a specified terminating number. The duration of the dial tone given the customer is between 2 and 30 seconds. The specified terminating number can be a maximum of 32 digits long.

During dial tone, the subscriber can use the line as a POTS line or for custom calling operations. Warm Line is not compatible with automatic lines, coin lines, multiparty lines, manual lines, or denied originating lines.

# Line signaling

The following paragraphs briefly describe the types of signaling (both subscriber to DMS-10 switch, and DMS-10 switch to subscriber) that may be used on subscribers' lines.

## Dial pulsing

Dial pulsing is a means of transmitting digit information as a series of momentary loop openings. The DMS-10 switch accepts a nominal rate of 10 pulses per second (pps).

## Digitone

Digitone signaling is a method of pushbutton dialing from the subscriber's station. Digitone signals use the voice path. The dial translates each digit into a combination of two tones: one tone from a high-frequency group and one tone from a low-frequency group. Details on Digitone frequencies can be found in the NTP entitled *System Performance Specifications* (297-3601-180).

## Optional ±130 V coin collect/return

The standard coin control voltage for DMS-10 the switch is +130 V dc for coin collect and -130 V dc for coin return. Through DMO, this can be reversed to -130 V dc for coin collect and +130 V dc for coin return.

## Loop disconnect immediate

This feature allows the 0-dB miscellaneous line pack (NT2T44) and Type B line card (NT6X18) to provide disconnect signal supervision to indicate that the calling party has disconnected. The disconnect signal is sent on NT2T44 or NT6X18 packs that are configured with ground start (prompt STRT = GND, in Overlay CPK (LPK) prompting sequence), or with loop disconnect start (prompt STRT = LPDS, in Overlay CPK (LPK) prompting sequence), capability. The disconnect signal is produced by opening the tip relay contacts to remove the talk ground from the line for 800 ms. This action removes talk current from the connected PBX or customer premises equipment. The disconnect signal occurs immediately following receipt of the on-hook from the calling party.

## Loop disconnect after disconnect timing

This feature allows loop disconnect to be provided through the cutoff relay on NT6X17 and NT6X18 line cards. Disconnect occurs after disconnect timing, configured in response to the DNCT prompt in the Overlay CNFG (CRTM) prompting sequence. This feature can be assigned on an office-wide basis, as a station option.

# 0-db lines

0-dB line packs are designed to eliminate the 2-dB nominal insertion loss of earlier circuit packs without any loss of circuit stability on intraoffice calls. The following packs are available:

- General Line (NT2T43) (one or two parties)
- Prepay Coin Line (NT2T45)
- Miscellaneous Line (NT2T44)
- Superimposed Ringing Line (NT2T67)
- Single-party Line (NT2T69)
- Eight-party, Multifrequency Ringing (NT2T75)

In order to provide the 0-dB feature, these packs must be used in conjunction with a Peripheral Processor pack (NT2T46) installed in any position on a Dual PE shelf (Positions 1 through 14).

# Cutoff strap

Before LCE, RSLE, or RSLM equipment is cut into service, the tip and ring terminals of the line circuits must be separated from the subscriber loops. The command CUT OVER LCEB/RSEB b/ALL in overlay PED activates the cutoff relay on the LCE, RSLE, or RSLM line cards. After a 128-ms "settling out period," a message is sent to the E99 CODEC to deactivate the relay, and the cutoff strap on the back of the shelf supplies current to keep the relay activated. While the cutoff strap supplies current to keep the relay activated, the installer completes the wiring and prepares to bring the LCE, RSLE, or RSLM into service. When the equipment is ready for cutover, the installer removes the cutoff strap from the back of the shelf. For information about the PED overlay, see NTP 297-3601-506, entitled *Maintenance Diagnostic Input Manual*.

# Coin Pad Enable/Disable

When a 0+ or 0- call has been placed from a coin station, the DMS-10 switch applies an *operator release* condition (-48 volts to ring and ground to tip) to the coin station, which enables the Digitone pad for further use and places the coin station totalizer in the *local* mode. After the call is routed to an operator, an *operator attached* condition (+48 volts to ring and ground to tip) is applied to the coin station line, which disables the Digitone pad (to prevent it from being used to simulate coin deposit tones) and places the totalizer in the *toll* mode so that the coin deposit can be sent to the operator.

When a 1+ call has been placed from a coin station, the DMS-10 switch applies an *operator attached* condition to the coin station line when the operator is connected, which disables the Digitone pad and places the totalizer in the *toll* mode. After the operator has verified the charges for the initial period, and prior to releasing the call, an *operator release* signal is sent to the DMS-10 switch, which then enables the coin station Digitone pad and places the totalizer in the *local* mode. The pad and totalizer remain in this state until an overtime or end-of-call seizure occurs.

Operator coin disposal control, that is, treatment for coin collection and coin return, totalizer control, and Digitone pad activation, can be applied only on a system-wide basis. The Coin Pad Enable/Disable feature allows this control to be applied on a per route basis.

The following conditions apply to the Coin Pad Enable/Disable feature:

- This feature applies only to routes which are used with operator traffic and use either inband or multiwink type signaling for receiver coin control.
- This feature is not applicable to semipostpay type coin stations.

For a procedure used to set up this feature, see SOP 0134 in NTP 297-3601-311, *Data Modification Manual*.

# Section 5: Trunk features

## Functional trunk types

The DMS-10 switch interfaces with the following functional trunk types.

### Dial toll switch

A dial toll-switch trunk carries incoming traffic from a class 4 office to the DMS-10 class 5 office.

### Toll switch

A toll-switch trunk carries incoming traffic from an operator to the DMS-10 office.

### Centralized Automatic Message Accounting

A Centralized Automatic Message Accounting (CAMA) trunk is a toll-connecting trunk that carries traffic from the DMS-10 switch to a class 4 office equipped with CAMA facilities. This trunk also carries ANI information for identification of the calling party.

### Recording-completing

A recording-completing trunk carries outgoing traffic from the DMS-10 switch to an operator.

### Extended Area Service (EAS)

The EAS trunk carries traffic between the DMS-10 switch and another class 5 office.

### Local tandem

A local tandem trunk carries traffic between two class 5 offices that is destined for a third office. The DMS-10 switch can interface with either an incoming or an outgoing local tandem trunk.

### Verification

A verification trunk is an incoming trunk to the DMS-10 office from either a toll office or an operator. It provides an operator access to the DMS-10 switch in order to verify the status of a subscriber's line.

### Recorded announcement

An optional Digital Recorded Announcement (DRA) Trunk pack (NT2T85) provides a single recorded message. The maximum message length is 16 seconds; however, the message length can be changed by adjusting the DIP-switch settings on the pack.

This pack has four ports; therefore, four subscribers can listen to the message simultaneously. The message can be played up to three times, as specified through data modification order (DMO).

> ***Note:*** *The DRA pack (NT2T85), combines the functions of both the analog recorded announcement unit and its associated Two-Wire E&M Analog Trunk pack (NT2T21).*

## Physical trunk types

The following paragraphs describe the physical trunk types that can interface with the DMS-10 switch. For detailed trunk interface information, see the NTP entitled *Circuit Interfaces for Lines, Trunks, and Test Trunks* (297-3601-184).

### Analog

The DMS-10 switch can interface with the following types of analog trunks:

- four-wire, E&M signaling trunks
- two-wire, E&M signaling trunks
- loop trunks

### Digital

The DMS-10 switch can interface directly with 1.544 Mb/s (DS-1) digital carrier facilities. Two of these facilities are the Digital Carrier Module (DCM) and the Digital Signal Interface (DSI). Detailed information on the DCM and DSI is found in the NTPs entitled *General Description* (297-3601-100) and *Equipment Identification* (297-3601-150). Signaling on digital trunks is defined as either “E&M” or “loop.”

## Virtual trunk types

The DMS-10 switch can interface to carriers on the internet using virtual trunks. These virtual trunks use SIP signalling and connect to the PSTN via the Packet Gateway Interface developed for the VoIP packet trunking feature.

## Signaling

The following paragraphs describe the types of signaling that can be sent or received by the DMS-10 switch.

## Multifrequency (MF) outpulsing

The DMS-10 switch is equipped to outpulse MF signals up to a maximum of 32 digits plus key pulse (KP) and start pulse (ST) signals. Each MF pulse consists of a combination of two frequencies.

## MF receiving

The DMS-10 switch receives up to 32 digits within 1.5 percent $\pm$10 Hz of nominal frequency.

## Dual-tone multifrequency (DTMF) outpulsing

The DMS-10 switch can store and outpulse DTMF signals (up to 16 digits) at a nominal rate of 7.5 pps. Each multifrequency pulse consists of a combination of two frequencies. The pulse time is 67.5 ms, and the interdigital time is 67.5 ms. This service is provided for extended area service (EAS) trunk routes only.

## DTMF receiving

The DMS-10 switch receives up to 16 DTMF digits within 1.5 percent $\pm$2 Hz of the nominal frequency. This service is provided for extended area service (EAS) trunk routes only.

## Dial pulse (DP) outpulsing

The DMS-10 switch can store and outpulse up to 16 digits at a nominal rate of 10 pps. Interdigital time determined by hardware is 400 to 1200 ms; the average time is approximately 1 s.

## Fast DP outpulsing

Fast DP outpulsing is the same as DP outpulsing, but the interdigital time can be programmed to a range of 300 through 900 ms.

## DP receiving

The DMS-10 switch accepts pulses for up to 14 digits within the range of 5 through 20 pps (nominal 10 pps).

## Immediate dial

The DMS-10 switch can receive or transmit DP on an immediate-dial trunk.

## Delay dial

When an incoming delay-dial trunk is seized, the called DMS-10 office returns an off-hook signal within 300 ms. When a receiver is connected, an on-hook signal indicates readiness to accept the digits.

## Wink start

When an incoming wink-start trunk is seized, the called DMS-10 office indicates that it is ready to receive digits by sending a 256 ms (nominal) off-hook signal.

## Overlap outpulsing

To reduce the delay between completion of subscriber dialing and the end of outpulsing by the DMS-10 switch, outpulsing may begin as soon as enough digits are received to determine and establish the outgoing route. However, an operating company-defined delay may be added on a per-route basis. Because outpulsing is faster than normal subscriber dialing, this delay ensures optimal outpulsing to the distant office.

## KP ST Trunk Signaling

The KP ST Trunk Signaling feature enables the DMS-10 switch to be used as a concentrator of direct access lines (DALs) by using a modified ANI as an authorization code for other vendor's switches.

Only MF trunks will be used for direct access lines. Normal overlap outpulsing as it exists for Equal Access will be available for the IDAL route.

## Integrated Services Digital Network Signaling User Part (ISUP)

The DMS-10 supports ISUP signaling, provided it is equipped and configured with Common Channel No. 7 (CCS7). Refer to NTP 297-3601-100 General Description for information regarding the CCS7 system. ISUP signaling is a method of providing information regarding call setup and release between switching systems over links, separate from the trunks being used to carry the setup/release and minimize post dial delay. ISUP signaling is transparent to the subscriber. However, due to the ability to carry more information then the traditional inband signaling, additional interoffice features are available to subscribers, such as CLASS and ISDN.

DMS-10 ISUP is capable of interworking ISUP with inband signaling. In other words, incoming inband signaling can translate to outgoing ISUP signaling and vice versa.

DMS-10 ISUP supports the ability to perform continuity testing, either on the outgoing side of a call, or the incoming side of a call. The continuity testing is provided over the voice trunks, ensuring a quality voice connection. Voice trunks not passing continuity testing are removed from call processing service, tested routinely in a background process, and maintenance personnel alerted.

DMS-10 ISUP supports intraLATA call setup, release, and interworking as specified by Telcordia specification GR-317. The interLATA call setup, release, and interworking is specified by Telcordia specification GR-394. All ISUP messages, formats, and protocol are defined in these Telcordia specifications.

## Session Initiated Protocol (SIP)

The DMS-10 supports SIP signaling over the virtual trunks associated with the SIP trunking feature. All SIP signaling is defined in RFC-3261 and RFC-3398 (excluding ISUP encapsulation).

# Trunk groups

The DMS-10 switch can accommodate a maximum of 511 incoming trunk groups and 511 outgoing trunk groups. A two-way trunk group uses one incoming and one outgoing trunk group.

## SIP Packet Trunk Groups

Trunk groups using the SIP signaling method use virtual trunks to interface to the SIP User Agent. Special trunk groups are defined to administer existing trunk group parameters and specific parameters for SIP signaling. All SIP Packet Trunk Groups are defined as two-way trunk groups. For more information see the NTP entitled *DMS-10 North America Data Modification Manual* (297-3601-311).

# Section 6: System features

## Local tandem

The DMS-10 switch can be used as a local tandem to route a local call originating in another office and destined for a third office.

## Toll tandem

This feature allows the DMS-10 switch to perform as a 4X or 4P class office.

## Automatic intercept system (AIS) Interface

The flexibility of the DMS-10 translation structure allows direct interfacing with AIS equipment.

## Improved Mobile Telephone Service (IMTS) interface

The DMS-10 translation structure allows the system to receive calls from and direct calls to IMTS equipment.

## Dialing

The DMS-10 switch supports the following dialing capabilities.

### Seven-digit local dialing

The DMS-10 switching system handles standard (seven-digit) local dialing.

### Direct Distance Dialing (DDD)

The DMS-10 switch provides the facility for DDD billing through LAMA, CAMA offices with Bellcore AMA formats, TOPS, NT 400/500, or AMR.

### International Dialing (IDDD)

An international number consists of a country code (CC) and a national significant number (NSN). Since the country code may be from 1 to 3 digits long, the national significant number consists of any digits in the international number that are not part of the country code. The international number may be from 7 to 15 digits long.

The maximum number of digits that can be dialed for an outgoing call, including the prefix, carrier code, operator digits, EBS or IBS prefix, and the called number, is 32.

#### Expanded International Dialing (EID) feature

Because the maximum number of digits in an international number is 15, a special billing module must be appended to the billing record to hold a terminating international number when it exceeds 12 digits in length. If the Expanded International Dialing Billing (EIDB) feature bit is set (see the FEAT prompting sequence of Overlay CNFG in NTP 297-3601-311, *Data Modification Manual*) when the terminating international number exceeds 12 digits in length, this module is appended to the billing record; if this feature bit is not set, the number is truncated to 12 digits for billing purposes. If the terminating international number exceeds 15 digits in length when the EIDB feature bit is set, the number is truncated to 15 digits for billing purposes.

#### Access codes

The following access codes are used to provide IDDD capability:

- for person-to-person calls, 01 + 7 through 15 digits
- for station-to-station calls, 011 + 7 through 15 digits
- for IDDD operator, 010

### Direct inward dialing / direct outward dialing (DID/DOD)

For private branch exchange applications, the DMS-10 switch provides direct inward dialing (DID) and direct outward dialing (DOD) capability.

### Office-wide ten-digit dialing enhancement

Public utility commissions (PUC) in several states have mandated that ten digits must be dialed for all calls, including local non-toll calls and intra-LATA or inter-LATA toll calls. To meet the requirements of this mandate, telcos may have removed seven-digit translations from their switches. The Office-wide ten-digit dialing enhancement enables the Integrated Business Services (IBS)/Enhanced Business Services (EBS) features, Busy Transfer (BTF), Don't Answer Transfer (DAT), IBS Intercom Dialing, and Directory Number Hunt Group overflow DN (DNH OVDN), which currently operate using seven-digit translations, to operate instead using ten-digit translations.

The enhancement is set up by responding YES to the new prompt, OTEN in the SYS prompting sequence of Overlay CNFG. When prompt OTEN is set to YES, these IBS/EBS features use ten-digit translations when they are invoked. When prompt OTEN is set to NO, the features use seven-digit translations when they are invoked.

## Access codes

The following paragraphs briefly describe the typical access and service codes supported by the DMS-10 switch and programmable by the operating company.

### Toll access

Toll access can be provided by dialing the digits 0, 1, 112, or any other access code defined by the operating company.

### Operator

The operator can be accessed by dialing 0 or any other access code defined by the operating company.

### 911 emergency

A common emergency service center can be accessed by dialing 911 or any other access code defined by the operating company.

> ***Note 1:*** If a 911 call originates on a Wireless line and the wireless subscriber goes on-hook, the interface DN is then released and made available. In addition, the emergency service bureau operator is disconnected and is unable to re-connect to the wireless subscriber.
>
> ***Note 2:*** When a 911 call is placed by a Wireless subscriber, the switchhook status for the call is lost and emergency ringback is also not available.

### Information codes

The DMS-10 switch can provide access to information services with access codes defined by the operating company. Typical information-access codes are:

- local 411
- HNPA 1-411 555-1212 1-555-1212
- NPA 1-NPA-555-1212 1-NPA-411

### Service codes

The DMS-10 switch provides flexibility in service-code assignment by allowing the operating company to define all service codes in the office data base.

## Local Coin Overtime

Coin calls, depending on their destinations, may have overtime charging. Such coin calls have an initial pay period and overtime pay periods. Coins are collected 30 seconds before the end of the pay period. At the end of the pay period, the Local Coin Overtime (LCO) feature performs a coin-presence test. If no coins are present, the feature connects both parties to a recorded announcement that instructs the coin-station user to deposit more coins. If more coins are deposited, an overtime period is begun.

Only Dial Tone First (CDF) and Coin First (CCF) coin phones are supported by the LCO feature.

### Local Coin Overtime / custom calling interface

The Local Coin Overtime / Custom Calling Interface feature enables parties called from a coin phone to utilize the Custom Calling and Custom Local Area Signaling Services (CLASS) features that require the parties to perform switchhook flashes during an established call. These features include Three-way Calling, Call Waiting, Call Hold, User Transfer (residential or business), Automatic Call Back, Automatic Recall, Calling Number Delivery, Customer Originated Trace (station or office-wide), and Screen List Editing. The feature enables the DMS-10 switch to continue processing LCO functions (coin presence tests, coin collection, and resource management) for coin parties involved in calls using these features, even when the coin party is in a temporary held state. When a coin party is connected to an LCO recorded announcement, all parties in the established connections are able to hear the announcement; if the coin party is on hold when connected to the LCO recorded announcement, only the coin party is able to hear the announcement.

## Operator services

The following paragraphs describe the operator services capabilities of the DMS-10 switch.

### Operator verification

This feature gives the operator access to any subscriber's line in order to verify the status of the line.

> ***Note:*** *All operator verification calls placed either to ISDN interface DNs or to Voice over IP (VoIP) Session Initiation Protocol (SIP) gateway line DNs are always routed to BVRF (busy verification) call treatment, regardless of the line state (busy or idle).*

### Operator-trunk coin control

This feature allows the operator to control coin-collect and -return functions on coin stations served by the DMS-10 switch.

### Ringback denial on multiparty lines

This feature prevents the operator from ringing back (for a 0+ or 0- call) on a multiparty line when the originating party cannot be identified. Reorder tone will be sent to the operator when an attempt is made to ringback on an unidentifiable line.

### Digitone Call Screening

The Digitone Call Screening feature allows the DMS-10 switch to screen calls based on whether the call is made from a digitone phone or a dial pulse phone. This screening determines whether operator assistance is needed for a customer to make a credit card call.

When a credit card call is made, the routing of the call is based on the key pulse signal sent before the called number, which indicates a digitone phone or a dial pulse phone. If the call is made from a digitone phone, the call is routed to an announcement informing the caller to enter the calling card number. If the call is made from a dial pulse phone, the call is routed to an operator who must ask for the calling card number.

# Carrier group busy

This feature uses the carrier group alarm (CGA) signal(s) from a carrier system to busy-out trunks associated with that carrier group. The following actions occur when a CGA is detected:

- All idle trunks associated with the carrier group are made busy.
- Trunks in use, associated with the carrier group, are forced to disconnect and are then made busy.
- When the CGA scan point is enabled at a remote unit, a trunk group specified by the operating company is taken out of service.

When the CGA clears, the associated trunks are returned to service. Detailed information is found in the NTP entitled *Data Modification Manual* (297-3601-311).

# Protection Line Switching

A *protection line* is an optional DS-1 (T-1) line that can be configured in conjunction with the primary DS-1 lines between the SCM-10S and the remote terminal (RT) housing the SLC-96 equipment. The protection line is switched in if one of the primary lines fails and is used as a direct replacement to maintain full service for the primary line.

Transmission failure on a primary line due to data, framing, or synchronization loss is eventually seen by the RT as a signal loss. The subscriber subsequently loses voice/data connection and experiences a dead line unless a switch can be made to the protection line.

Protection Line Switching can be handled either automatically or manually. When the switch occurs, calls in the ringing or talking state are maintained and calls being processed are dropped.

## Manual switching.

With manual switching, operating company personnel must input a command requesting that a switch be made to the protection line. A separate command must be input to release the protection line and return service to the primary line. See overlay DED in NTP 297-3601-506, *Maintenance Diagnostic Input Manual*, for these commands.

### Automatic switching.

With automatic switching, the protection line is activated only when a failure occurs on a primary DS-1 line or when it is taken out of service. Either the host DMS-10 switch or the RT can automatically initiate the switch to a protection line. In the event of excessive faults, the end that detects the faults triggers the switch. Once the faults have been cleared, the system automatically switches back to the primary line.

Because it handles system messaging between the DMS-10 switch and the RT, the DS-1 line for shelf A has priority over the DS-1 lines for shelves B, C, and D when multiple protection switch requests occur simultaneously. Otherwise, protection switch requests are handled on a first-come, first-served basis. If two or more protection switch requests occur simultaneously, the lines are addressed in alphabetical order (that is, B, C, D).

#### Protection Line Switching limitations

The following limitations apply to Protection Line Switching:

- The S/DMS AccessNode configured as a SLC-96 does not support Protection Line Switching.

#### Swact back

A new *Swact back* feature allows the XPM to switch back to the original active controller if the new active controller has trouble soon after a switch of activity. The new active controller tests for two way communication with the host. If the communication test fails, control is returned to the original active controller. If the test passes, then the original active controller is reset and returned to service to stand by. If a Swact back occurs, the fault must be fixed.

## Loop Modularity

This feature allows for modular growth and shrinkage of existing DS-30A and SRI Remote loop interfaces. This feature is implemented by the MOVE prompting sequence in overlay CNFG and the LCMC prompting sequence and RSLC prompting sequence in overlay NET. Loop Modularity also permits the unbundling of RLCMs at one site, using the MOVE prompting sequence, in order to assign each RLCM its own RMM and its own site, which is necessary for the RLCM Emergency Stand-Alone (ESA) feature.

## Equal Access Services

Equal Access Services allow the DMS-10 switch to comply with the Modification of Final Judgement (MFJ), which is the agreement reached between the U. S. Department of Justice and AT&T in 1982. According to the MFJ, the service provided by the Bell Operating Companies (BOCs) for all long-distance carriers must be equal in type, quality, and price. This equality of service must also be provided by non-BOC offices within specified time schedules.

The extent of Equal Access Services provided depends on the feature group that is configured in the DMS-10 office. The DMS-10 switch may be configured with Feature Group A, B, C, or D. Feature Groups A, B, and C are interim Equal Access configurations, and Feature Group D complies completely with the MFJ.

## Equal Access network

As part of the Equal Access plan, the switching network is divided into Local Access and Transport Areas (LATAs). Each LATA is a geographic area within which an operating company may offer its telecommunications services. Both intra-LATA traffic, which originates and terminates within the LATA, and inter-LATA traffic, which originates in one LATA and terminates in another LATA, are supported. Within each LATA, there may be one or more Equal Access End Offices (EAEO) and Access Tandem (AT) offices. The DMS-10 switch may be an AT or, if Feature Group D is configured, an EAEO.

With Equal Access, the switching network has a three-level hierarchy. The lowest level is the EAEO, which provides complete Equal Access Services in addition to the usual end office function of station loop terminations for connection to other loops and to trunks. The next level is the AT office, which concentrates and distributes intra-LATA and inter-LATA traffic. The highest level is the long-distance carrier, which is a 950 carrier, intra-LATA carrier, inter-LATA carrier (IC), international carrier (INC), or consolidated carrier that performs two or more functions.

Each long-distance carrier is assigned a carrier access code that may be dialed by the subscriber or added to dialed digits by the system software. The carrier access code is a set of four digits, which is part of the 101XXXX or 950-XXXX code that may be used by subscribers to place calls by way of the indicated carrier. X can be any number from 0 to 9.

## Equal Access DMS-10 software configuration

Feature Groups A and D must be configured on a per-office basis in the DMS-10 switch. Overlay CNFG may be queried through the SYS prompting sequence to determine if these groups are configured. Overlay EQA is used to define long-distance carriers (CARR prompting sequence) and country codes (CC prompting sequence). The overlays and prompting sequences are in the NTP entitled *Data Modification Manual* (297-3601-311).

## Interim Equal Access plans

The type of Equal Access Services provided depends on the feature group that is configured in the DMS-10 switch. Feature Groups A, B, and C provide partial Equal Access Services.

## Feature group A

Feature Group A (FGA) serves as an interface between the DMS-10 switch and the customers who purchase FGA service. Typical FGA customers are long-distance carriers other than AT&T, private line resellers, and private networks. FGA provides the FGA customer with a line-side termination plus a seven-digit access code (the directory number of the access line) at the DMS-10 office.

By dialing the seven-digit FGA number, subscribers of an FGA customer that is served by the DMS-10 switch (or by another end office in the same local calling area) can call an FGA line that terminates at the DMS-10 switch. Upon recognizing the seven-digit FGA number, the DMS-10 switch completes the call to the FGA line and applies ringing. The FGA customer answers the originating call by returning off-hook and dial tone to the calling subscriber, who must use Digitone signaling to input any further dialing information required by the FGA customer (usually a personal identification number) and the called number.

For originating (to the FGA customer) and terminating (from the FGA customer) calls, Bellcore format AMA records are generated by the DMS-10 switch configured for FGA. For originating FGA calls, the AMA record starts when the FGA customer answers and ends when the call is disconnected. For terminating calls, the AMA record starts when the called party answers and ends when the call is disconnected. These AMA records of FGA line usage are used by the operating company to charge the FGA customer for access and can also be used by the FGA customer to bill its subscribers.

The following conditions apply to FGA:

- The FGA option must be assigned to each FGA line through overlay DN.
- The following call types can be restricted when terminating in the DMS-10 switch from an FGA line: N11, 0+/0-/01+, 101XXXX), and 950.
- FGA numbers must have Directory Number Hunting configured through overlay HUNT to collect originating FGA overflow counts.
- With Directory Number Hunting, only one number is listed for billing purposes.
- Subscriber automatic calling number identification (ANI) is not provided to the FGA customer.
- Dual-tone multifrequency (DTMF) or dial pulse signaling is allowed on FGA lines for terminating FGA calls.
- Because the DMS-10 switch is transparent to address-signaling information between the calling party and the FGA customer, the calling party must use Digitone signaling.

### Feature group B

Feature Group B (FGB) is an interim Equal Access Service that allows a non-EAEO to provide its subscribers with access to inter-LATA and international carriers. In the DMS-10 switch, FGB is the same as FGA except that the FGB customer has carrier access instead of line-side access and Bellcore format AMA records are generated by the end office or the AT.

### Feature group C

Feature Group C (FGC) is an interim Equal Access Service that is the same as Feature Group D, except that conventional signaling (KP + called number + ST + KP + ANI + ST) is used.

## Final Equal Access plan

Complete Equal Access Services are provided by Feature Group D.

### Feature group D

Feature Group D (FGD) allows the DMS-10 switch to be used as an EAEO or AT office. This ability includes the provision of network control signaling, carrier access codes, automatic calling number identification, answer supervision, operator service signaling, and the Bellcore format AMA information to bill carriers for access charges.

DMS-10 subscribers can access an IC or INC by dialing the prefix of the final Equal Access Plan. This uniform dialing plan allows DMS-10 subscribers to access all carriers on an equal basis.

This prefix takes the form 101XXXX, where XXXX is the carrier access code; the four-digit carrier access code makes available 9999 possible codes. The DMS-10 switch will prefix a leading zero to all three-digit carrier codes entered so that the codes will be in the four-digit format. When it is necessary for one of these modified three-digit carrier access codes to be outpulsed over a three-digit outgoing trunk, the 0 will be stripped from the code.

The following conditions apply to FGD:

- Remote Call Forwarding (RCFW) calls cannot be forwarded outside World Zone 1, except for 10-digit North American Numbering Plan calls to Mexico.
- LAMA is required in a DMS-10 switch configured for FGD.
- IC operator-assisted 0- calls and INC operator-assisted 101XXXX 0/00 calls are routed to the inter-LATA operator.
- Only three-digit (NPA) ANI is provided for IC/INC calls made on a multiparty line with no automatic party identification.
- DMS-10 switches can address up to 512 carriers. DMS-10 switches can address up to 256 toll regions.

- A PIC may be assigned to eight- and ten-party flat rate lines. All parties on the line must be assigned the same PIC.
- During the transition period, traditional signaling may be used for inter-LATA calls, but it is restricted to MF only (wink start or delay dial). DP signaling and immediate start are not allowed for inter-LATA calls. For a tandem connection using traditional signaling, the AMA record is generated in the EAEO when the first wink is returned from the AT. Because this first wink is not the true carrier-connect signal from the IC, compensation may be required by the operating company to adjust the true elapsed time from carrier connect.
- Up to eight EOAT trunk groups may be assigned.

The following features are not supported by FGD:

- Terminating Subscriber Line Usage Study (SLUS)
- Tracer Records for AMA
- Short Supervisory Transitions (SSTs)
- Network Management
- Interface to No.2 Service Evaluation System (SES)
- Machine-Detected Interoffice Irregularities
- Input/Output Error Messages
- Operator Recall Service

### Subscriber abbreviated dialing

Three abbreviated dialing plans, AD1, AD2, and AD3, are offered with FGD. AD1 allows the subscriber to use abbreviated dialing codes for calls being routed by an IC or INC. AD2 and AD3 allow the subscriber to use Speed Calling for calls being routed by an IC or INC. Speed Calling is a generic term that refers to Short Speed Calling (SSC), Long Speed Calling (LSC), Convenience Dialing (CVD), Convenience Dialing Controller (CVDC), Group Speed Calling (GSC), and Group Speed Calling Controller (GSCC).

If a subscriber has been assigned one of the Speed Calling options, an Equal Access prefix 101XXXX, and directory number may be associated with a Speed Calling code.

**Abbreviated Dial 1** Cut-Through Dialing, or AD1, allows the subscriber to route a call to an IC or INC by dialing the Equal Access prefix, 101XXXX, followed by an octothorpe, which indicates the end of dialing. The subscriber then dials an abbreviated dialing code or other information, depending upon the prompt returned by the IC or INC. DTMF dialing capability is required to use AD1. AD1 is optional for each IC or INC.

**Abbreviated Dial 2** AD2 allows the subscriber to route a Speed Calling call through an IC or INC by dialing the Equal Access prefix, 101XXXX, followed by a Speed Calling code. The digits corresponding to the dialed Speed Calling code are outpulsed to the IC or INC represented by the carrier access code. The subscriber does not need to enter the Equal Access prefix before the Speed Calling code if the subscriber chooses to use his PIC.

Because AD2 is used in conjunction with Speed Calling, a Speed Calling option must be assigned to the subscriber's line. The subscriber must assign a directory number to the Speed Calling code, as explained in the “Residential Services” section of this NTP. DTMF capability is not required for AD2.

**Abbreviated Dial 3** AD3 provides compliance with the LSSGR standard for Equal Access. AD3 allows the subscriber to route a Speed Calling call through an IN or INC by dialing a Speed Calling code that has been associated with an Equal Access prefix, 101XXXX, and a directory number. The digits of the directory number corresponding to the Speed Calling code are outpulsed to the IC or INC represented by the carrier access code.

Because AD3 is used in conjunction with Speed Calling, a Speed Calling option must be assigned to the subscriber's line. The subscriber must assign an Equal Access prefix and a directory number to the Speed Calling code, as explained in the “Residential services” section of this NTP. DTMF capability is not required for AD3.

## LEAS Route Access

Local Equal Access System (LEAS) Route Access allows the DMS-10 switch to interface with class 4 offices that provide the LEAS feature to non-conforming end offices. In order to interface with the class 4 offices, the DMS-10 switch must have the capability to collect and outpulse up to 32 digits in one continuous stream. The LEAS route is used to outpulse these digits.

The subscriber may access a long distance carrier by dialing a 1+ or the equal access prefix (101XXXX), followed by the 7 to 15 additional digits required to complete the dialing sequence. For 1+ calls, office automatic number identification (ANI) data are used to determine the presubscribed carrier. For 101XXXX calls, the 101 digits indicate that the call is a LEAS type, and the XXXX is used to process the call to a specified carrier at the class 4 office.

## Exchange Access Operator Service System Signaling (EAOSS)

EAOSS provides the ability to combine 1 + coin traffic with other types of traffic on a single trunk group. This feature handles signaling between an Equal Access End Office (EAEO) and an Access Tandem (AT) associated with an Operator Services System (OSS) or IC/INC (interLATA carrier / international carrier). Other types of traffic with which OSS traffic may be combined include:

- telephone company operator service traffic and telephone company traffic not requiring operator services
- both IC traffic requiring telephone company operator exchange access services and IC traffic not requiring telephone company operator exchange access services
- both INC traffic requiring telephone company operator exchange access services and INC traffic not requiring telephone company operator exchange access services
- calls between an EAEO and IC/INC routed through an AT that require special functions, such as coin control, ringback, and hold

When EAOSS is implemented, an IC/INC may arrange for the telephone company to provide operator system exchange access assistance on particular types of calls. These calls will then be routed to the TCOS (Telephone Company Operator Service) before being routed to the IC/INC. EAOSS must also permit special operator functions, such as hold, recall, and expanded in-band signaling, on certain types of calls for IC/INCs that wish to provide their own operator services. Therefore, EAOSS will provide:

- all traffic types between an EAEO and an AT to be combined on a single trunk group
- the option of employing a new signaling protocol to distinguish the types of calls that are being transferred over the trunk group, based on the type of call and type of dialing equipment the originator may be using
- traffic on the combined trunk group to be marked as requiring special handling when an off-hook is received from the AT instead of an acknowledgment wink
- 0+ interLATA or 01+ international calls dialed from a non-presubscribed line to be routed to the TCOS if 101XXXX is not dialed (on an office option basis)
- 1+ interLATA or 011+ international calls dialed from a non-presubscribed line with special service classes to be routed to the TCOS if 101XXXX is not dialed (on an office option basis)
- 0+ interLATA or 01+ international calls dialed from a presubscribed line to be routed first to the TCOS if 101XXXX is not dialed (on an office option basis)
- the ability to assign specific Numbering Plan Area (NPA) codes where directory listing service (DLS) calls will be routed to the TCOS
- intercept calls to be routed to an AT with TCOS
- calls scheduled to be handled by TCOS will not be billed. A billing record will be generated for carrier connect time calculation but the call will be marked as a free call

# Coin Box Accounting for Revenue Allocation (CBA-RA)

Coin Box Accounting for Revenue Allocation (CBA-RA) enables the telco to record coins collected by a coin telephone, allowing coin box revenues to be allocated to the carriers serving the coin telephones in any given exchange.

When an operator-attached signal is received from an Operator Services System (OSS), indicating that coins must be collected, a Universal Tone Receiver (UTR) pack (NT4T02) or Network Interface pack (NT8T04) channel is attached to the coin phone. This attachment provides OSS connection for monitoring the coin deposits. Monitoring is then terminated when an operator released signal is received from the OSS. The monitoring is performed each time the operator returns to the call for coin collection, including the initial collection and all subsequent overtime collections.

# Operator Services Network Capability (OSNC)

OSNC allows the use of the Integrated Services Digital Network User Part (ISUP) of Signaling System No. 7 (SS7) to support operator services. Operator services include toll and assistance services (e.g., calling card) and listing services (e.g., directory assistance and intercept). The DMS-10 can serve as an originating, intercepting, tandem, or terminating office in an SS7 Operator Services System (OSS) connection.

## Originating Operator Services

The DMS-10 may serve as an originating equal access end office (EAEO) for an OSNC call. The procedures for originating OSS connections support signaling functions such as the establishment and release of the connection, updates of the end-to-end connection information, connection hold, coin control, and network recall (flash). The OSS connection to the DMS-10 end office may be a direct trunk connection or it may be tandemed through one or more intermediate switches.

### Modified NOA vs. Basic NOA

Before OSNC it was difficult to combine operator services and non-operator services calls on the same trunk group (TG). OSNC introduces the concept of a Modified NOA (Nature of Address), which differs from Basic NOA in the rules surrounding the encoding of the Called Party Number (CDN) parameter NOA field in the IAM for 1+ dialed calls that also require operator services such as coin, hotel, and restricted originated calls.

For Basic NOA, the CDN parameter NOA is encoded to represent the dialed access prefix for the call. Calls dialed as 0+ or 0- have a NOA value that includes an operator requested indication, while calls dialed as 1+ do not. Since there are certain 1+ dialed calls that require operator services even if the subscriber did not dial the 0+/0- prefix, it is difficult for the OSS to separate traffic requiring operator services from non-OSS traffic if combined on the same TG.

For Modified NOA, the operator services CDN parameter NOA code points are modified to reflect the need for operator services even if the dialed prefix is not 0. The Operator Services Information (OSI) parameter is also introduced to provide explicit information about the access prefix that was dialed (0, 01, 1, or 011). This scheme allows the CDN parameter NOA field to be encoded to reflect operator services requested on 1+ dialed and other no prefix dialed calls (for example, 411).

For both the Basic and Modified NOA options, the Service Activation Parameter (SAP) is included in the IAM. The feature code indicator field of the SAP is encoded to specify whether the originating office can process Connection Hold requests from the OSS.

### IXC Involvement in Operator Services Calls

An Inter-Exchange Carrier (IXC) may be involved in an OSNC call. IXC involvement is determined during translation of the dialed number based on digits dialed and/or pre-subscription options for stations, thousands groups, and/or EBS groups. If the Modified NOA signaling scheme is used, the DMS-10 checks the carrier’s data to determine whether the IXC accepts Access Signaling information, and if so, includes it in the OSI parameter of the IAM.

### Combining OS and Non-OS traffic on the Same Trunk Group

The DMS-10 enables different types of traffic to be shared on the same ISUP trunk group, as follows:

- All types of operator services (OS) connections may be combined on the same TG, as long as those connections support a common ISDN bearer capability.
- OS connections may be combined on the same TG with non-OS traffic when the ISUP signaling for the call uses the Modified NOA field option.
- When ISUP signaling for the OS call uses the Basic NOA field option, OS connections other than Intercept connections and 1+ screened connections may be combined on the same TG with non-OS traffic.

### Coin Control

Coin control is a function performed by operator services to control both the flow of coins through the coin phone and the state of the keypad. The primary mechanism for sending coin signals provided by the OSNC feature is the Facility (FAC) message containing the Service Activation Parameter (SAP). sent from the OSS to the originating office. Coin control signals may also be receive in a SAP in the ACM (Address Complete Message) sent from the OSS. The following coin control signals may be received in a SAP: Coin Collect, Coin Return, Network Service (operator) Attached, and/or Network Service Released.

### Connection Hold

Connection Hold, also known as operator hold, allows the operator to maintain control of the call even after the calling party has gone on-hook. It provides a means to preserve the connection facilities when the calling party disconnects.

The mechanism for establishing Connection Hold is the SAP in the IAM. The originating office offers Connection Hold by encoding “hold available” in a Feature Code Indicator of the SAP. To invoke Connection Hold, the OSS responds with another SAP parameter, encoded as either “hold request” or “hold request, with acknowledgment”, in either the ACM or any FAC message. Connection Hold cannot be established on ISDN subscriber lines or on intercept connections and is only available on OSNC calls involving an IXC if the carrier data is set to allow it.

When Connection Hold has been established, the OSS controls when the connections are released. The originating DMS-10 reports on-hooks and off-hooks to the OSS by sending FAC messages containing a SAP. The OSS evaluates the services required on the call and determines if the connection should be retained or released.

### Removing Connection Hold

The originating DMS-10 recognizes Connection Hold release requests, encoded as either “hold release request” or “hold release request, with acknowledgment” in the Feature Code Indicator field of the SAP of any FAC, ANM, or CPG message received from the OSS.

### Calling Party Disconnect with Connection Hold in Effect

When Connection Hold is in effect on the originating connection, the DMS-10 does not release the connection when the calling party goes on-hook. When an on-hook is detected, the DMS-10 sends a Facility message with a SAP encoded “Disconnect Request”, places the call in a Disconnect Request Pending state, and waits for the OSS to request the next action. Additionally, the DMS-10 starts a hold timer based on the timer setting for the route (1, 2, 3, or 4 minutes).

While the OSNC call is being held, the OSS sends FAC message with a SAP encoded to indicate Connection Hold Continuation Request to the DMS-10 every 20 seconds until the OSS requests the next action. Upon receipt of this FAC, the DMS-10 resets the hold timer.

If an off-hook indication is received from the calling party while the call is in a Disconnect Request Pending state, the DMS-10 cancels the hold timer. If the DMS-10 receives a REL message from the OSS while the call is in a Disconnect Request Pending state, normal release procedures are followed.

### Calling Party Reconnect with Connection Hold in Effect

If the calling party goes back off-hook while in a Disconnect Release Pending state, the DMS-10 sends a Facility message with a SAP encoded to the OSS to indicate a reconnect request, removes the call from the Disconnect Request Pending state, cancels the hold timer, and waits for the OSS to request the next action.

### Maintenance Call Clearing

Maintenance Call Clearing (MCC) is performed on OSNC calls in the Disconnect Request Pending state if the OSS has not released a connection that has been idle for a specified length of time. If a FAC message with SAP encoded “Connection Hold Continuation Request” is not received from the OSS office before the hold timer expires, the DMS-10 releases the call and prints a trouble message.

### Calling Party Ringback Request

A ringback request is a means for the OSS to cause the calling party’s phone to ring when further dialog is required. Connection Hold must be established and the connection must be in a Disconnect Request Pending state in order for the ringback request to be honored. A ringback request is signaled to the originating office using a FAC message containing a SAP with a Feature Code Indicator field encoded “Ringback Request”. When the DMS-10 receives a ringback request, it initiates alerting on the calling party’s line.

### Network Recall (Flash)

Network recall is a means for the calling party to request re-connection to an operator after the call has been set up. Connection Hold must be established in order for the network recall to be available. A network recall is requested by the calling party by initiating a flash on their phone. The flash triggers a FAC message containing a SAP with a Feature Code Indicator field encoded “Network Service Recall” to be sent from the DMS-10 to the OSS.

## Intercept Operator Services

A calling user may attempt a call to a line that is no longer in service or has trouble on the line. Such a call may invoke operator services, e.g., intercept, that provides information on why the call cannot be completed and/or provides the correct number. The need to establish intercept service is determined through translations of the called party’s line information. Before OSNC, SS7 signaling could not distinguish between the different types of intercept calls. These intercept call types are used by the OSS to select announcements used to inform the subscriber of the new number.

The intercept call type indication is sent by the originating DMS-10 in the IAM containing a SAP. Intercept originations are only without IXC involvement and should not be routed to IXC networks. Additionally, Connection Hold is not permitted on intercept calls.

## Tandem Operator Services

For the most part, a tandem switch in an OSS connection operates the same as for basic call control and does not need to recognize an OSNC call as such. One exception is the following: If the DMS-10 is serving as a tandem office in an OSS connection not involving an IXC (that is, a connection to a LEC OSS). and an IAM is received with Called Party NOA set to “no address operator” and no Called Party digits, the DMS-10 will translate on the digits “00” to allow a “0-” or “00” call to translate successfully and be routed out of the office. Translations should be set up at the tandem office to allow the digits “00” to route to the LEC OSS.

Other than this one exception, a tandem switch in an operator services (OS) call operates the same as for basic call control, and as such, will simply pass ISUP messages between the OSS and the originating switch in the backward and forward directions without modification.

If the DMS-10 serves as a tandem office for an OSNC connection and the incoming OSNC IAM translates to or overflows to a route using inband signaling, special OSNC operator services (such as Connection Hold) will not be available on the call since it is not ISUP all-the-way, which is required for OSNC treatment. Similarly, if a the DMS-10 serves as a tandem office and receives an incoming signal from an inband trunk for an operator call, there are no special interworking requirements, and no conversion of the MF signals to OSNC messaging is performed if the call translates to an ISUP route.

## Terminating Operator Services

The DMS-10 can be the terminating end office of an OSS call over an incoming SS7 connection. As a terminating end office, the OSS call termination will appear to be a normal ISUP call with the calling party as the originating party. There are no special OSNC terminating procedures at the DMS-10.

## OSNC administration

The OSNC feature requires ISUP signaling and Equal Access be enabled in the office. It is controlled by an OSNC feature bit. Refer to Service Order Procedure (SOP) 0242 in NTP 297-3601-311 *Data Modification Manual* to set up routes, carriers, and trunk groups for OSNC.

### Assigning an ISUP Operator Signaling System route

The existing ISUP/IEQA route type is used for OSNC traffic by setting the OPR prompt to “YES” in overlay ROUT. Additional values for the Connection Hold timer and coin functions can also be set for the OSNC route. The Secondary route type (STYP) for the route may be set to any of the existing STYPs, including EOAO.

If a call with a dialing pattern indicating operator services are requested translates to a route without the operator route indication (OPR) set, the functionality provided for the call will be consistent with basic ISUP call control with no special OSNC functionality provided.

### Assigning IXC information

On a per-carrier basis, the following data can be assigned in overlay EQA: whether the IXC receives access signaling information for OSNC calls, whether the IXC honors Connection Hold requests, and whether the Basic NOA Field Option or the Modified NOA Field Option should be used for OSNC calls using this carrier.

### Assigning trunk group information

On a per-trunk group basis (outgoing and two-way ISUP TGs only), the following data can be assigned in overlay TG: whether operator traffic may be carried on the trunk group and whether the Basic NOA Field Option or the Modified NOA Field Option should be used for OSNC calls using this trunk group.

If a call with a dialing pattern indicating operator services are requested translates to a TG without the operator traffic indication set, the functionality provided for the call will be consistent with basic ISUP call control with no special OSNC functionality provided.

### Assigning a TCOS for an HNPA

On a per-HNPA basis, a Telephone Company Operator System (TCOS) carrier can be assigned in overlay AREA. This carrier will be used for an OSNC call that translates to a route with STYP = EOIC, EINC, EAIC, or EAIN when 1010XXX dialing is not used and the originator is not presubscribed to an IXC.

## Flash resolution

The establishment of a second call leg (for example, 3WC, UTF, EBS CHD) while the originator is involved in an OSNC call and Connection Hold is available for activation by the operator is not allowed. Instead of providing dial tone in response to a subscriber’s flash, the operator recall signal is provided on an OSNC call with Connection Hold established. If Connection Hold is available on the call but has not yet been activated by the OSS, a subscriber’s flash will be ignored.

If a DMS-10 subscriber with 3WC, UTF, or EBS with CHD is active on a non-operator call, flashes, and dials the operator, an OSNC call can be established as the second call leg; however, Connection Hold is not available for this call.

## Automatic Callback (ACB) interaction

ACB cannot be performed for OSNC calls from a DMS-10 station.

## CALEA interaction

If an originating subscriber is in Connection Hold on an OSNC call and the call is monitored by CALEA (Communication Assistance for Law Enforcement Agencies), on-hook, off-hook, and ringing indications are sent in Call Data Channel messages.

## Call Forwarding interaction

A call cannot be forwarded to an OSS using OSNC signaling for any forwarding feature. The applicable DMS-10 forwarding features are: Call Forwarding (CFW), Call Forward Busy (CFB), Call Forward Don't Answer (CFD), Call Forward Fixed (CFF), and Call Forward Remote Access (CFRA).

## Call Waiting (CWT) interaction

CWT is not allowed on a subscriber’s station when Connection Hold is available for a OSNC connection. If Connection Hold is not available for the OSS call (for example, if the originator is an ISDN phone or if the routed-to IXC does not accept Connection Hold requests), an OSNC connection can be call waited.

## Screening List Editing (SLE) interaction

A DN that translates to an ISUP/IEQA route using a dialing pattern requiring operator services cannot be added to an SLE list for any features utilizing these lists. The DMS-10 features using SLE lists are Selective Call Forwarding (SCF), Selective Call Rejection (SCR), Selective Call Acceptance (SCA), Selective Distinctive Ringing/ Call Waiting (SDR), and Simultaneous Ringing (SRNG).

## Automatic Message Accounting (AMA)

The introduction of SS7 signaling to support operator services does not cause a change in which AMA call type codes, structures, and modules are used. The only changes involve the triggers for starting and stopping AMA-related timers, and the use of the IXC Call Event Status field.

Normally the elapsed time from carrier connect is the time from carrier connection until an on-hook is detected from the originating line, the time an SS7 REL (Release) is received at the originating office, or the time an REL is sent by the end office, whichever occurs first. On OSNC connections, Connection Hold may have been invoked. If so, an on-hook signal from the originating line does not result in the immediate release of the circuit. Therefore, elapsed time from carrier connect does not end at the receipt of an on-hook. On OSNC calls with IXC involvement, if Connection Hold has been invoked on the connection, the originating DMS-10 determines the elapsed time from carrier connect as the time from carrier connection until the time the originating end office either receives a REL from the network or sends a REL to the network, whichever occurs first.

If Connection Hold is established at any time on an OSNC call with IXC involvement and an ANM (Answer Message) is not received, the DMS-10 records the value "05" in the IXC Call Event Status field in the IXC Delivery module of the AMA record.

# Presubscription (PRES)

Presubscription is an operating-company tariffed service that permits each EAEO subscriber's communications to be routed automatically (that is, without the subscriber dialing the carrier access code) to an IC or INC of the subscriber's choice. The selected carrier, which is assigned using the PRES station option, is the subscriber's primary interconnect carrier (PIC). The subscriber may also gain access to other ICs or INCs by using the appropriate carrier access code (101XXXX).

The PRES option may be assigned to a Remote Call Forwarding Appearance (RCFA) in lieu of entering "101XXXX" as a part of the Remote DN (RDN) for the RCFA. Entire thousands groups may also be presubscribed to an assigned carrier.

## Multiple PIC Option (MPO)

This feature enables a subscriber to select up to two additional presubscribed inter-LATA, intra-LATA, or international carriers. The DMS-10 switch determines, for inter-office calls, which one of the three possible presubscribed PICs to use based upon the call type and toll region routing used for that call, based on a DEST *x* SCRN *y* translations action. RCFAs and entire thousands groups may also be assigned multiple PICs using the PRES, PRS2, and PRS3 options.

### Intraoffice MPIC

The Intraoffice MPIC feature enables the DMS-10 switch also to determine which of the three presubscribed PICs to use for intraoffice toll calls, based on a THGP *x* SCRN *y* translations action.

## Secondary Carrier Screening (SCS)

SCS provides the operating company with an alternative to the carrier (CARR) screening test defined in translations. With SCS, the operating company can create up to five screening translators to be selected for use by the DMS-10 switch based upon the region routing for a call.

### Presubscription for Intra-LATA Calling (EPA-PICL)

Presubscription for Intra-LATA Calling (PICL) allows a subscriber (or an RCFA) to access an intra-LATA carrier without dialing the carrier access code 101XXXX. The station/RCFA must be presubscribed to a primary interconnect carrier (PIC), the PICL station option must be assigned, and the call must be routed to the TRAP toll region. If the station/RCFA is not presubscribed or the PICL option is not assigned, a non-101XXXX call will be routed according to the SCRN *nn* in the translations line. Refer to the NTP entitled *Data Modification Manual* (297-3601-311) for information on assigning station options and toll regions.

### Presubscription Indication (PSI)

Presubscription Indication (PSI) provides the IC or INC with an indication of whether a calling station is presubscribed to the carrier or the subscriber has gained access to the carrier by dialing the Equal Access prefix, 101XXXX, for that carrier. The key pulse signal that precedes the calling number is used to identify a call where a subscriber has obtained access to a carrier by dialing 101XXXX. The key pulse signal (KP or KPP) is assigned through overlay ROUT, ROUT prompting sequence.

PSI treatment for service access code (SAC) calls (that is, N00 calls) depends on whether the originator is billed for the call. If the originator is not billed for the call, for example, 800 call, the call is treated as though the subscriber is presubscribed to the carrier. If the originator is billed for the call, for example, 900 call, the call is treated as described in the preceding paragraph.

## Inter-LATA Restriction (IRST)

A subscriber with the IRST option assigned cannot access carrier routing for Abbreviated Dial 1 (AD1) calls, 950 calls, and 101XXXX.

## Specific Carrier Restriction (CRST)

A subscriber with this option is denied the ability to make calls by way of specific inter-LATA or international carriers; the calls are routed, instead, to an IRST (inter-LATA restricted) generic condition. The subscriber may access the specific carrier(s) only to make operator or non-billable service access code calls (for example, 800 calls). The subscriber's PIC can be one of the specified carriers.

When a carrier is to be deleted from the office and the carrier is associated with the CRST option on any station, that option must be deleted from all stations before the carrier is deleted. For more specific information about deleting a carrier, refer to the NTP entitled *Data Modification Manual* (297-3601-311).

### Multiple Selective Carrier Denial (MSCD)

This feature is an enhancement to the CRST option. When MSCD is configured in the office, the maximum number of carriers to which one subscriber's access is restricted is increased from 2 to 512.

## Specialized IDDD ticketing

An enhancement to FGD allows the DMS-10 switch to be used as an IC switch. To support this ability, international calls received from an incoming carrier and translated to an outgoing carrier can be assigned an ANI number for the purpose of generating an AMA billing record. To set this feature in software, see overlay TG, prompting sequences INC and/or 2WAY, in the NTP entitled *Data Modification Manual* (297-3601-311).

## Code Confliction (EQA-CDC)

Call translation confliction may occur when the DMS-10 switch routes seven-digit calls on an alternate route through an access tandem (AT) office that is outside of the call originator's Home Number Plan Area (HNPA). The confliction occurs when the AT is in a Number Plan Area (NPA) other than the call originator's HNPA and the AT uses an inter-LATA carrier (IC) or international carrier (INC) in that NPA. The alternate IC or INC may not receive enough digits to route the call to the originator's HNPA and may attempt to route the call to its own HNPA.

The Code Confliction (EQA-CDC) feature prevents this translation conflict by prefixing the HNPA of the originator to the called digits. The HNPA prefix will be added only when the alternate route is to an AT in a NPA different from that of the call originator, when seven digits are dialed, and on routes that are configured for such prefixing. Refer to overlay ROUT (ROUT prompting sequence) in the NTP entitled *Data Modification Manual* (297-3601-311) for route configuring information.

For EQA-CDC, the call originator is defined as the subscriber who originates the Equal Access portion of the call. For example, when Subscriber A dials Subscriber B and the call requires the use of a carrier, Subscriber A is the call originator. However, when Subscriber A dials Subscriber B and Subscriber B's line is call forwarded over a carrier, Subscriber B is the call originator.

## N11 carrier access (EQA-N11)

Subscribers may access the customer services offered by an intra-LATA, inter-LATA, or international carrier. Access is obtained by dialing the carrier's Equal Access prefix, 101XXXX, followed by N11, where N is a number from 2 through 8. “911" calls are routed according to the SCRN *nn* in the translations line, usually to the operating company's emergency service bureau, and are not routed according to the carrier's SCRN *nn*. If the carrier associated with the dialed access code does not provide customer services, the call is routed to a generic route (AD1I, INCC, IRAI, IRST, IERI, or INLI). If the subscriber dials only N11, the call will be routed according to the SCRN *nn* in the translations line, usually to the operating company's customer service center. The operation of EQA-N11 is not affected by a subscriber's decision to choose a primary interconnect carrier (PIC).

## Subscriber dialing

The digits that must be dialed by a subscriber are determined by the type of carrier, the call type, and whether the subscriber's inter-LATA or international primary interconnect carrier (PIC) is used. The required dialing for inter-LATA, intra-LATA, and international calls is listed in Tables 6-A, 6-B, and 6-C, respectively. Parentheses indicate optional digits.

**Table 6-A:**
**Equal Access inter-LATA carrier subscriber dialing**

| Call Type | PIC Not Used | IC PIC Used |
|---|---|---|
| Customer service from carrier | 101*XXXX* + *N*11 | 101*XXXX* + *N*11 |
| Customer service from operating company | *N*11 | *N*11 |
| Cut-through to carrier (AD1) | 101*XXXX* + # | 101*XXXX* + # |
| Local operator assisted 0- | 0- | 0- |
| Non-operator assisted | 101*XXXX* (+1) + 7/10 digits | 1 + 7/10 digits |
| Operator assisted 0+ | 101*XXXX* + 0 + 7/10 digits | 0 + 7/10 digits |
| Operator assisted 10*XXX* 0- or 00 | 101*XXXX* + 0 | 00 |

***Note:*** *N = service digit, XXXX = carrier access code, and # = end-of-dialing indicator.*

**Table 6-B:**
**Equal Access intra-LATA carrier subscriber dialing**

| Call Type | Digits Dialed |
|---|---|
| Customer service from carrier | 101*XXXX* + *N*11 |
| Customer service from operating company | *N*11 |
| Non-operator assisted | 101*XXXX* (+1) + 7/10 digits |
| Non-operator assisted with presubscription for intra-LATA calling configured | 1 + 7/10 digits |

***Note:*** *N = service digit, XXXX = carrier access code, and # = end-of-dialing indicator.*

**Table 6-C:**
**Equal Access international carrier subscriber dialing**

| Call Type | PIC Not Used | INC PIC Used |
|---|---|---|
| Customer service from carrier | 101*XXXX* + *N*11 | 101*XXXX* + *N*11 |
| Customer service from operating company | *N*11 | *N*11 |
| Non-operator assisted | 101*XXXX* + 011 + *CC* + *NSN* (#) | 011 + *CC* + *NSN* (#) |
| Operator assisted 0+ | 101*XXXX* + 01 + *CC* + *NSN* (#) | 01 + *CC* + *NSN* (#) |
| Operator assisted 10*XXX* 0- or 00 | 101*XXXX* + 0 | 00 |
| Within World Zone 1, but outside continental United States | 101*XXXX* (+1) + 10 digits<br>or<br>101*XXXX* + 0 + 10 digits | (1+) 10 digits<br>or<br>0 + 10 digits |

***Note:*** *CC is the country code, N is the service digit, NSN is the national significant number, XXXX is the carrier access code, and # is the end-of-dialing indicator (octothorpe). The CC and NSN may be 7 to15 digits long.*

## Dual Integrated Modem

This feature provides an interface to the DMS-10 switch for the purpose of remote TTY access. It is a self-contained replacement for the existing dual SDI/modem system arrangement, consisting of a main circuit board (NT3T93), two Bell 212A compatible modem cards, and a paddleboard with cable. It eliminates the need for standalone or rack-mounted dial up modems, modem shelves, and associated ac powering.

The Dual Integrated Modem supports 300 baud or 1200 baud rates and automatically adapts to the speed of the calling modem. The modem pack is an auto-answer, auto-dial unit which recognizes Hayes commands. The DMS-10 switch uses Hayes commands to communicate with the Integrated Modem; however, operating company personnel cannot input these commands at a terminal.

The modem functions in two modes, command and data. The command mode is used for initial connection and diagnostics. In command mode an incoming data call will cause the modem to go off hook and connect with the calling modem. After this it enters data mode, whereby data is exchanged between the host and remote. The modem remains in data mode until the calling modem goes off line, at which time it detects loss of carrier and go to an on hook condition.

The Dual Integrated Modem pack is assigned using DMOs to define the pack type and location. For more information, refer to overlay CNFG (LOGU prompting sequence) in the NTP entitled *Data Modification Manual* (297-3601-311).

## Emergency service bureau (ESB)

The DMS-10 switch is required to provide the capability of routing emergency calls to a designated ESB. With this service, the calling party originates an emergency call by dialing 911 or any three- or seven-digit number set up in the office translations.

When both the originating line and the ESB are served by the same switching system, the following features are available:

- Bureau Hold
- Forced Disconnect
- Switchhook Status of Calling Party
- Emergency Ringback
- Disconnect Tone/Bureau Integrity Check

In order to provide these features, the Central Office must interconnect with Western Electric 8A Key Telephone System (KTS) equipment at the ESB. These features, however, are not required when the calling party is served by a customer-premises switching system; only one-way service, terminating at the ESB and using loop-type ringdown operation, is required.

In some installations, a trunk interfacing with a switchboard can be used. The switchboard may be dedicated to emergency reporting, or trunks may appear on an operating company switchboard.

Expanded 911 service is not provided by the DMS-10 switch; however, the DMS-10 switch can transmit address digits and ANI information over toll trunks to offices fitted with the expanded 911 feature package.

Requirements for the outgoing toll trunk at DMS-10 end offices are the same as for the trunk to TSPS or CAMA.

The DMS-10 switch may have a direct connection to the ESB or it may route emergency calls, by way of tandems, to the ESB. In the latter case, calls are routed with EAS traffic or over dedicated trunks. However, if the agency involved requires bureau hold, switchhook status, and ringback features, a direct connection must be provided. The DMS-10 switch may also act as the tandem switch for 911 service.

## Direct dedicated lines

Where a dedicated connection is used from the originating local office direct to the ESB, 20-Hz ringing (continuous or interrupted, depending upon the ESB ringing type in the Configuration Record) is applied toward the ESB, and audible ringing is applied toward the calling party. When the ESB answers, the ringing is tripped without sending a reversal toward that calling party, so that no charge is made for the call. After the ESB attendant answers, the connection is maintained as long as the attendant is off-hook, even if the calling party hangs up. However, if the attendant hangs up, and the calling party does not, the connection is released within a specified time period. This delay reduces false disconnects due to hits or improper operation of the ESB.

## Tandem trunking

When a local message Network circuit is used from the originating office to a tandem office, the requirements of the terminating circuit at the tandem office are the same as those for the local office direct connection, except that no provision need be made for forced hold. The method of forcing a disconnect is necessarily different, because the tandem connection is held by the incoming trunk. The terminating circuits time for a specified period (same as that for forced disconnect timing) on receipt of an on-hook (following a confirmed answer) from the agency. At the expiration of on-hook signal, an off-hook is again maintained for a specified period and then removed. This sequence activates the called party timed disconnect feature in the originating office and releases the entire connection. In the process, the calling party is charged, if it is a coin or message-rate line, but this is an acceptable limitation.

## Bureau hold

In cases where the calling party does not properly identify himself/herself, the ESB can hold the established connection to identify, with the help of the operating company, the calling line. This hold can be maintained regardless of the action of the calling party. The bureau hold feature can also be used to identify the calling line when a harassing call has been made to the ESB.

## Forced disconnect

With the Forced Disconnect feature, the ESB can release the connection regardless of the action of the calling party. This feature is essential to prevent a caller from jamming incoming circuits. It can also be used to release legitimate calls, when the calling party fails to hang up on completion of a call, a situation that occurs more frequently with emergency service than with exchange or toll service.

## Switchhook status

This feature allows the ESB to determine if the calling party has gone on-hook after the connection has been established to the ESB, even if the bureau has placed the call on hold.

## Emergency ringback

With Emergency Ringback, the ESB can hold a line open and ring back the calling party who has gone on-hook or who has gone away from an off-hook handset. This feature is useful when the calling party has failed to provide all of the necessary information to the ESB before he/she hangs up or when such information must be verified.

***Note:*** *In the event that the calling party made the 911 call from a multiparty line, the ESB rings back all ringing codes on the line.*

## Disconnect tone / bureau integrity check

This purpose of this feature is twofold: it enables the ESB to verify the integrity of 911 trunking facilities and alerts the ESB when a call has been abandoned.

If the ESB should go off-hook and seize the 911 trunk without prior seizure by a calling party, 120 IPM reorder tone is returned to the ESB for the duration of the seizure. If this seizure persists for more than a predetermined interval (128 ms - 155 s), a minor central office alarm is raised. The alarm is retired by the ESB going back on-hook.

If a calling party abandons a 911 call before the ESB answers, a 120 IPM reorder tone is received by the ESB.

## Group alerting

Group Alerting is provided in the DMS-10 switch through the use of an external Tellabs 291 Fire Reporting and Conference System.

## RSC-S Emergency Service Bureau

Analog and digital ESB trunks are supported in the RSC-S.

### RSC-S ESB operation in Host mode (not in ESA mode)

The operating company can define ESB trunks subtending the RSC-S and can configure translations to route calls to terminate on an ESB trunk. The ESB trunk is a dedicated one-way, outgoing-only trunk.

When a 911 call is placed, the call terminates on the ESB trunk group. If all trunks in the group are busy, an alternate route set up by the operating company is used. After the ESB attendant answers the call and a conversation path is established, the call is released only upon receipt of a disconnect signal from the service bureau. If a disconnect is received by the ESB from the calling party following answer, a steady low tone is applied to the ESB until either the calling party returns to off-hook status, a disconnect signal is received from the ESB, or a ringback flash signal is received from the ESB.

### RSC-S ESB trunk operation in ESA mode

RSC-S ESB trunks support the following features when the RSC-S is in ESA mode:

- Free call
- Bureau call
- Forced Disconnect
- Calling party hookswitch status (the RSC-S sends low tone to the ESB to indicate that the calling party has gone onhook)
- ANI outpulsing (ANI is not spilled for calls originating from a multiparty line in ESA mode)
- Emergency ringback (for a call originating on a multiparty line, the first ring party receives the ringback even when the originator of the 911 call is one of the other parties)

Maintenance cannot be performed on the trunks when the RSC-S is in ESA mode.

# Nailed-Up Connections

The Nailed-Up Connections feature allows the establishment and maintenance of a continuous voice path between any two voice ports. Any line or trunk is a valid source or destination for a Nailed-Up Connection, provided lines are assigned a directory number or trunks are part of a trunk group. A Nailed-Up Connection should remain intact through SYSLOADs and Initializations and once established, it should be monitored for faults and outages.

A total of 256 connections are available per office (both cluster and non-cluster). Nailed-Up Connections, assigned in overlay ROUT, eliminate the need for external channel banks, additional span lines and channel drop equipment for special applications. Information on the overlay prompting sequence is contained in the NTP entitled *Data Modification Manual* (297-3601-311).

The DMS-10 supports ISDN B-channel data and D-channel packet Nailed-Up connections. Both ISDN Nailed-Up connections are assigned through Overlay ROUT. An office supports up to 256 B-channel connections (BCON) and 128 D-channel connections (DCON).

## Common Channel Signaling #7 (CCS7)

Common Channel Signaling #7 is a platform for features that require data exchange through a CCS7 signaling network. The DMS-10 switch serves as an end office Service Switching Point, capable of generating and transmitting signaling information and other data over a CCS7 signalling network. See the NTP entitled *General Description* (297-3601-100) for information on CCS7 implementation.

### Enhanced 800 (E800) services

The E800 feature uses external database queries to obtain routing and billing information for 800 calls. E800 is available in Equal Access End Offices (EAEOs) configured with Common Channel Signalling #7 (CCS7).

When an 800 call is placed, a message is sent to the Service Control Point (SCP) in the CCS7 network. The SCP serves as the interface to the external database. If the message is sent successfully, it contains the following information:

- the calling party address, consisting of the Destination Point code (DPC) and subsystem number
- the called party address, consisting of a subsystem number, translation type, and global title translator

The DPC, subsystem number, translation type, and global title translator are defined through the DMOs. Once the database receives this message, the corresponding routing and billing information is returned to the DMS-10 switch through the SCP. The switch then determines the destination of the call: either an inter-LATA / international carrier, another office, a generic condition, or a recorded announcement.

The E800 service can be accessed by Enhanced Business Service (EBS) lines and by the following DMS-10 custom-calling features:

- Speed Calling - E800 numbers may be stored in the existing speed lists
- Call Forwarding - E800 numbers can be stored as the forwarded number if the 800 number does not terminate to an International carrier

- Remote Call Forwarding Appearance - E800 numbers can be defined for remote call forwarding appearances
- Call Waiting and Three-Way Calling - E800 calls can be the second or third member of a three-way call. Hookflashes are recognized once the call is in a talking state
- INWATS - E800 calls terminating to an INWATS line within the office will be completed

## Multiple E800 LATA enhancement

The originating LATA is included in database queries sent to the SCP. The Multiple E800 LATA enhancement enables the DMS-10 switch to serve multiple LATAs at the thousands group level. Thus, if a LATA has been assigned to the thousands group, it is included in the query sent to the SCP; if a LATA hasn't been assigned, the LATA specified for the office is sent. For a procedure used to set up the E800 feature and the Multiple E800 LATA enhancement, see SOP 0020 in NTP 297-3601-311, *Data Modification Manual*.

## 800 AT (Access Tandem) services

The Enhanced 800 services (E800) feature (see the description above) only supports queries to Service Control Points (SCP) if the originator of the call is a line; thus, queries to SCPs from incoming trunks are rejected. The 800 AT Services feature enables the DMS-10 switch acting as an Access Tandem / Service Switching Point (AT/SSP) to support queries to SCPs on incoming trunks from subtending Equal Access End Offices (EAEO), non-EAEOs, and tandem offices (see Figure 6-1). The incoming trunk signaling supported by the feature may be either inband or Integrated Services Digital Network User Part (ISUP) which conforms to technical requirements described in Bellcore documents TR-TSY-000317 and TR-TSY-000394.

> ***Note:*** *In order for the AT/SSP to support queries from subtending private branch exchanges (PBX), Improved Mobile Telephone Services (IMTS), and cellular systems, the line featured trunk patch is required.*

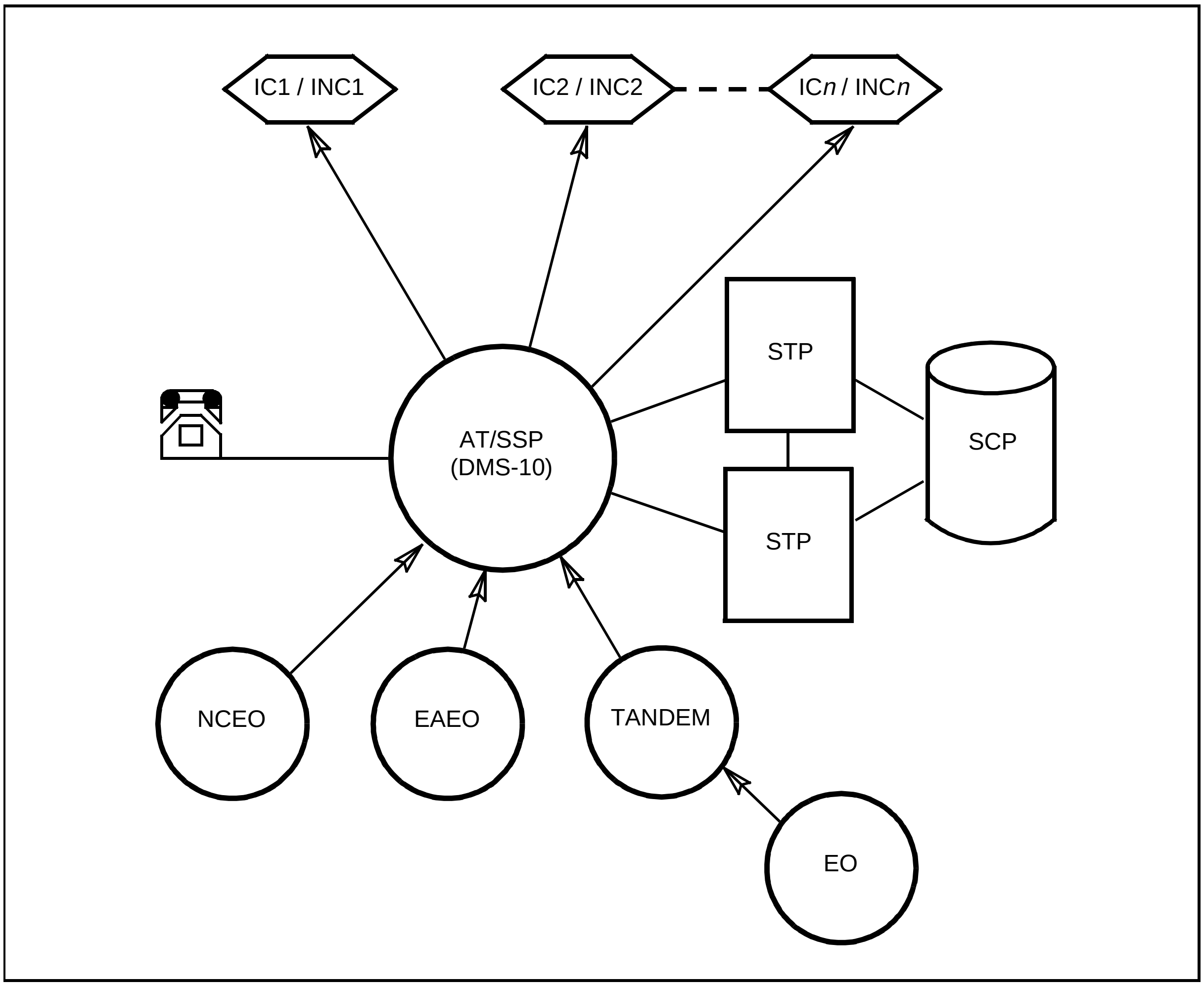
### Incoming signaling

If a subtending office is an EAEO, the AT/SSP recognizes Number Service calls by a special code, XXXX, in the incoming signaling stream, KP + 0ZZ + XXXX + ST. Number Service calls from non-EAEO or tandem offices are identified by the signaling streams, KP + 800 + NXX + XXXX + ST or KP + 00Y + NXX + XXXX + ST (the code, 00Y, specifies the NPA of the originating line if the subtending office is a non-EAEO or is acting as a tandem office serving more than one NPA). If the incoming trunk is an ISUP trunk, this information is contained in the IAM. If the ANI information is available, it is passed to the SCP; if it is not available, then either the NPA + NXX or NPA of the originator is passed to the SCP.

### Database query

The database query that the AT/SSP sends to the SCP in response to an incoming trunk call from a subtending office is identical to a query sent from an end office SSP. However, the number returned by the SCP is sent to a translator defined for the incoming trunk group rather than to a POTS translator. If the SCP returns a carrier and a routing number, then the codes, 0ZZ + XXXX, are prefixed to the routing number before it is sent to the translator; the code, 0ZZ, specifies that the call is a domestic Number Service call, while the code, XXXX, refers to the carrier returned by the SCP. If the returned number is an international number, then the code, 1NX, is prefixed instead of the 0ZZ code. If the call is to be routed via telephone company trunks, the routing number, as returned by the SCP, is sent to the translator.

**Figure 6-1: 800 AT Services architecture**



### Outgoing signaling

The AT/SSP supports both ISUP and inband trunks on the outgoing side of a call. The method of outgoing signaling used depends on the information passed to the AT/SSP by the SCP. If telephone company trunks are to be used, traditional signaling is used (even if Feature Group D signaling was used on the incoming trunk). If a carrier is specified, then either traditional or Feature Group D signaling is used, regardless of the method of incoming signaling. The 800 AT Services feature also supports ISUP interworking with inband signaling (for example, ISUP signaling used from the end office to the access tandem, or, MF signaling used from the AT/SSP to the carrier).

## Intermediate Tandem/Access Tandem (IT/AT) Link

This feature enables the DMS-10 switch to serve as an intermediate tandem (IT) between the EAEO and an access tandem (AT) when the EAEO doesn't have direct access to an inter-exchange carrier (IC). The IT delivers calls to the AT, which serves as the carrier's point-of-presence. This relationship is shown in Figure 6-1.

### ISUP signaling

The AT normally selects an outgoing trunk to an IC based on signaling information from the EAEO. In Multifrequency Feature Group D (MF FGD) signaling, this is encoded as 0ZZ + XXX(X), where the 0ZZ digits act as steering digits for an AT trunk group which connects to the IC specified the by the XXXX digits. This same information for CCS7 signaling is found in the Transit Network Selection (TNS) parameter of the IAM: the *circuit code* is the functional equivalent of the 0ZZ digits and the *carrier identification code* (CIC) is the same as the XXXX digits. Carrier connect timing at the EAEO begins with the receipt of either an acknowledgement wink for MF signaling or an Exit Message (EXM) for CCS7 signaling sent by the AT. In the IT/AT link configuration, the IAM sent from the IT to the AT contains the circuit code and CIC in the TNS received from the EAEO and the EXM received from the AT is forwarded by the IT to the EAEO.

### E800 SSP calls

When E800 SSP capability resides in the IT, the IT launches an EXM to the EAEO after the query to the SCP has been sent. The IAM sent from the IT to the AT contains the TNS parameter with the CIC received from the SCP. The circuit code contains the value that was set up during post query translations. The IT terminates the EXM from the AT and records carrier connect time.

## 800 Number Exhaust

The 800 Number Exhaust feature enables the DMS-10 switch to process new toll-free numbers introduced in response to the anticipated exhaustion of 800-NXX-XXXX numbers in the North American Numbering Plan. The new toll-free numbers will be made available through the use of additional, “interchangeable” NPAs, starting with “888.”

### Impact on E800 query response handling

E800 uses CCS7 signaling to query a database containing information about toll-free numbers. An E800 query is sent to a Signaling Transfer Point (STP) which, in turn, sends a request to a Service Control Point (SCP) for information about the toll-free number. The response to the query contains carrier information and, possibly, a change applying to the toll-free number dialed by the subscriber. If an alternate number is returned in the response, the number must be examined to see if it is toll-free, since toll-free numbers are not retranslated. Because up to eight INPAs may be defined, a list of valid toll-free numbers serviced by the switch must then be queried to determine whether the alternate number prefix (INPA-NXX) is valid.

The destination point codes (DPC) used for routing an E800 request for toll-free number information are obtained from stored memory through the toll-free number prefix. Two DPCs are assigned to each toll-free code; load sharing splits the usage between the two DPCs. The translation type codes associated with each DPC, which are used to identify the appropriate translation table at the STP, and the originating subsystem number, which is used to identify the subsystem at the SSP that is responsible for processing toll-free calls, are also obtained from memory through the toll-free prefix. These three network identification elements are defined by the operating company through the DMOs.

### Impact on Automatic Call Gapping

Automatic Call Gapping is a network management tool used to block calls to toll-free numbers that are either overloaded or unassigned. Prior to the 800 Number Exhaust feature, the toll-free numbers being monitored were stored without the “800" prefix. With the introduction of additional toll-free prefixes in the 800 Number Exhaust feature, the prefix must also be stored with the toll-free numbers being monitored. In addition, the maximum number of toll-free numbers that can be monitored simultaneously by the tool has been increased.

### Impact on Y-codes

Y-codes are used by end offices that handle multiple HNPAs. When the DMS-10 switch acts as an access tandem, a Y-code is sent to the carrier that handles the toll-free call in order that the originating NPA can be identified. Because the new toll-free numbers do not use Y-codes, the entire toll-free number must be sent to the carrier instead. Y-codes will continue to be used, however, for “800" prefix toll-free numbers.

## Vendor Digital Recorded Announcement (VDRA) unit

The Vendor Digital Recorded Announcement (VDRA) unit, provides the telco with the capability of providing announcements for customers subscribing to CLASS features. For a description of the CLASS features, refer to the section of this NTP entitled “Residential services”.

A VDRA unit contains a set of announcements that are pre-defined by the vendor. Depending on the model of VDRA unit, the unit may also provide the capability of recording new, non-CLASS announcements. This capability gives the telco the opportunity to change announcements or add new ones to satisfy office needs.

Multiple VDRA units can be connected to the DMS-10 switch. VDRA units interface with the DMS-10 switch through four-wire E&M trunks, through Digital Carrier Modules (DCM), or through Digital Signal Interfaces (DSI).

## Switched 56 kbps Services

The Switched 56 kbps Services feature enables subscribers to connect their computers and terminals through the DMS-10 digital network and standard DMS-100 family Datapath equipment consisting of Data Units (DU), Data Line cards (DLC), and Datapath Extension cards (DPX). The feature provides the operating company with the following capabilities:

- use of existing subscriber loops for integrated voice and data switching
- efficient use of trunks and lines (no dedicated facilities)
- multiple device access from data terminal equipment
- sharing of modems among users and trunks
- efficient use of computer ports through queuing and hunting features of the DMS-10 switch
- use of DMS-10 system and station features
- full-duplex asynchronous and synchronous data transmission at various user-selectable speeds (300-19200 kbps asynchronous, 1200-64000 kbps synchronous)
- use of standard, non-loaded two-wire subscriber loop for distances up to 5.4 km (18000 feet) on 22-gauge cable
- use of standard A- and B-bit signaling on the trunk side, and standard RS-232C and V.35 interfaces to subscribers' Data Terminal Equipment (DTE)

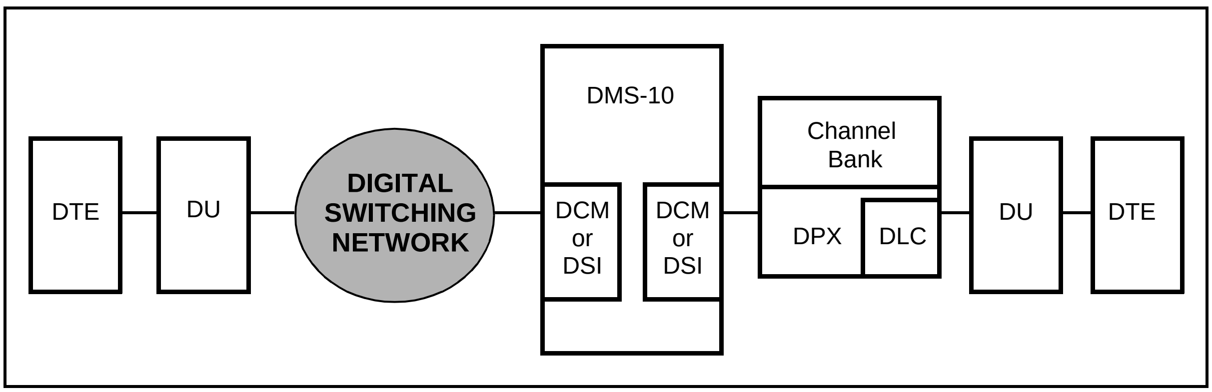
In the Switched 56 kbps Services feature configuration digital information is transported between the DMS-10 switch and the Datapath equipment over a DS-1 carrier. The link from the DU to the DMS-10 switch occupies one 64-kbps (DS0) channel on the DS-1 carrier. The link appears to the DCM or Digital Signal Interface (DSI) as an E & M trunk with A-bit signaling used to and from the DMS-10 switch for call supervision. The DPX acts as the intermediary in this process, occupying one slot on the channel and converting the DS0 signal provided by the channel bank into the format expected by a DLC, which separates data and signaling information. The format of the encoded user data on the DS0 channel conforms to T-link protocol.

For more information about the DMS-100 family Datapath equipment, consult the appropriate DMS-100 family NTPs.

## Subscriber access

Call origination is allowed when the originating subscriber's data terminal equipment (DTE) *data terminal ready* (DTR) lead is on. The subscriber presses the DN key on the Data Unit (DU) and, if the origination attempt is successful, hears a dial tone and dials the access number. The subscriber hears a ringback until the call is answered. When the call is answered, the Connect lamp on the DU flashes; after the T-link sync is found, the Connect lamp is turned on. A call disconnect occurs if the RLS key is pressed by the calling or called party or if the DTR lead is turned off. The DPX configuration is shown in Figure 6-2.

**Figure 6-2: Switched 56 kbps Services - DPX configuration**




If the Automatic Line feature is assigned to the line (AUT station option; see NTP 297-3601-311, *Data Modification Manual*, overlay DN [STN]), calls originated from the DU are routed automatically to a specified terminating number. There is no indication to the calling party, however, that the call is being routed automatically; the subscriber receives dial tone until the call is answered.

### Baud rate settings

The subscriber has control over baud rate setting. The baud rate of the originating DU is passed to the terminating DU when synchronization is first established using T-link end-to-end signaling protocol. This baud rate is used at both ends, regardless of the position of the baud rate selector on the terminating DU.

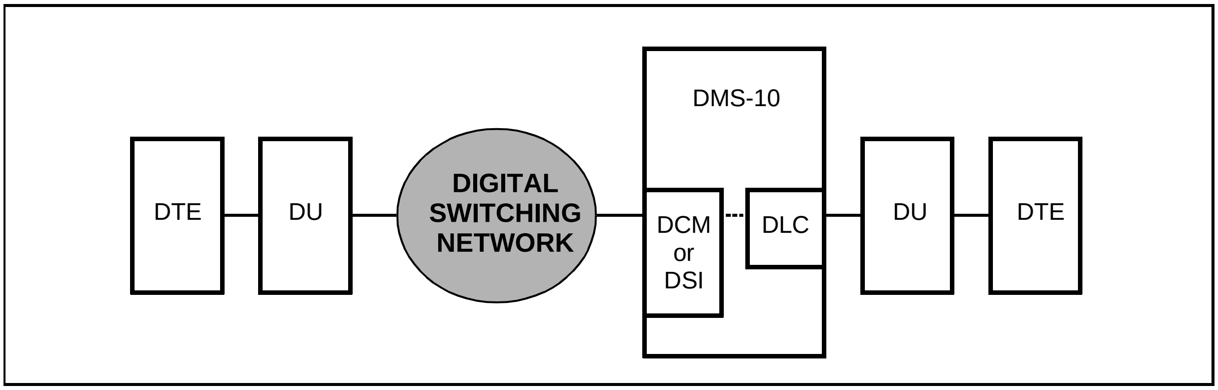
## Billing

Billing for the Switched 56 kbps Services feature is on a flat-rate basis. The subscriber, as call originator, is also billed for toll calls, regardless of the status of the second leg of a call.

## Datapath Line Card

The Datapath Line Card feature eases the implementation of the Switched 56 kbps Services feature both for operating companies and for subscribers by eliminating the need for DPX cards and dedicated DCM or Digital Signal Interface (DSI) channels to interface the DMS-10 switch with a data unit (DU). Instead, an NT6X71 line card provisioned in a DMS-10 LCM is connected directly to the DU (also called *Datapath terminal*) through a two-wire line. This is illustrated in Figure 6-3.

**Figure 6-3: Switched 56 kbps Services - Datapath Line Card configuration**




The Datapath Line Card feature also provides subscribers with access to other features through *Circuit Switched Digital Capability (CSDC)*, a generic digital service that allows data communication at 56 kbps. In addition, the feature provides for interworking with SL-100 and SL-1 PBXs.

The following conditions apply to the Datapath Line Card feature:

- A call between two DUs requires all-digital facilities over the telephone network. Devices that provide echo cancellation or suppression cannot be used.
- The maximum loop length between the DUs and the NT6X71 Data Line Card is 18000 feet for 22- or 24-gauge wire or 14000 feet for 26-gauge wire.
- Modem pooling is not supported.
- The only DU function keys supported are DN (simulation of off-hook signal), RLS (simulation of on-hook signal), and digit keys.
- Datapath calls on LCM-based remotes in ESA mode are not supported.

### Datapath Line Card testing

Datapath Line Card testing consists of Data Line Card (NT6X71) and DU diagnostics and the testing of transmission quality of the loop-extended datapath through Bit Error Rate Testing (BERT). For a description of BERT, see section 9 in NTP 297-3601-500, *General Maintenance Information*.

# NPA Split for CLASS

An NPA split occurs when the DNs in a given NPA are split between two NPAs. During a “permissive dialing period” following the NPA split, the dialing plan allows the use of either NPA when subscribers dial a DN in a new NPA. For most applications, translations can be adjusted to recognize both NPAs. But the CLASS features, Automatic Recall (AR), Automatic Call Back (ACB), Selective Call Forwarding (SCF), Selective Call Rejection (SCR), Selective Call Acceptance (SCA), and Selective Distinctive Ringing Call Waiting (DRCW), require that the dialed DN and the actual DN match. The NPA Split for CLASS feature enables these CLASS features to recognize both NPAs associated with a single DN.

## DN validation problems addressed by the NPA Split for CLASS feature

The following paragraphs describe typical DN validation problems that the NPA Split for CLASS feature addresses.

### AC or AR activated for duplicate DN

Subscriber A, who has the Automatic Recall feature, receives a call from subscriber B, whose DN is subsequently made part of a different NPA. Because subscriber B's DN at the time of the call and the new DN do not match, subscriber A will receive a denial announcement in response to an AR activation. The NPA Split for CLASS feature ensures that subscriber A's AR or ACB call is successfully placed to subscriber B, even though subscriber B's NPA has changed.

### Attempt to add a duplicate DN to a screening list

A subscriber who has a Screening List Editing (SLE) feature attempts to add a DN that has a new NPA to a screening list. Because the DN dialed by the subscriber does not now match the actual DN, the DN is not added to the list, and the subscriber receives an error announcement. The NPA Split for CLASS feature enables a DN to be validated even though the NPA associated with the DN has changed.

## Screening problems addressed by the NPA Split for CLASS feature

The following paragraphs describe typical screening problems that the NPA Split for CLASS feature addresses.

### Screening an AC or AR subscriber's DN

Subscriber A, whose NPA has changed, receives a call from subscriber B, who has Selective Call Acceptance (SCA). Subscriber A activates AC or AR. Because subscriber A's new DN doesn't match the DN on subscriber B's screening list, subscriber A will not receive a rejection announcement and subscriber B will receive the call. The NPA Split for CLASS feature ensures that subscriber A's call is screened, even though subscriber A's NPA has changed.

### Screening an original call

Subscriber A, whose NPA has changed, calls subscriber B, who has the Distinctive Ringing Call Waiting (DRCW) feature. Because subscriber A's new DN doesn't match the DN on subscriber B's screening list, subscriber B will not be notified about the call. The NPA Split for CLASS feature ensures that subscriber B is notified about subscriber A's call even though subscriber A's DN has a new NPA.

## Screening List Editing problems addressed by the NPA Split for CLASS feature

The following paragraphs describe typical screening list editing problems addressed by the NPA Split for CLASS feature.

### Deleting a DN from a screening list

If a subscriber tries to delete a DN that has a new NPA from a screening list, the DN dialed by the subscriber will not match the DN on the list and the subscriber will receive an error announcement. The Split NPA for CLASS feature ensures that the subscriber can delete a DN from a screening list even though the NPA for the DN has changed.

### Adding a DN to a screening list

Because the DN dialed by the subscriber and the DN on a the subscriber's screening list don't match after an NPA split, a subscriber can add a duplicate DN to the screening list. The Split NPA for CLASS feature ensures that a DN can be added to a screening list only once, even if the DN's NPA has changed.

## Feature operation

The NPA Split for CLASS feature enables up to 32 NPA split pairs, consisting of the former NPA and the new NPA, to be maintained. For each NPA split pair, up to four options may be assigned that determine how the NPA split pair interacts with the CLASS features:

- Option 1 ensures that, during TCAP DN validation, either NPA in the split pair is valid for a DN when the DN is added to a screening list, or when the DN is used for the Automatic Recall or Automatic Call Back features.
- Option 2 ensures that DNs associated with the NPA split pair can be screened with either the former or new NPA against entries on a screening list.

- Option 3 enables subscribers to delete a DN associated with an NPA split pair from a screening list.
- Option 4 prevents subscribers from adding duplicate DNs (that is, a DN that is already in the list, associated with another NPA) to a screening list.

For a Service Order Procedure used to configure the NPA Split for CLASS feature, see SOP 0120 in the NTP entitled *Data Modification Manua*l (297-3601-311).

# Digital PX Trunks

The Digital PX Trunks feature provides line type functions on digital, PBX, and mobile/cellular type 1 trunks. The feature also enables the DMS-10 switch, by way of a Digital Carrier Module (DCM) or Digital Signal Interface (DSI), to provide Foreign Exchange Facility Access (Foreign Exchange Originator [FXO] and Foreign Exchange Subscriber [FXS]) functions otherwise performed by channel banks.

The Digital PX Trunks feature consists of three parts: Foreign Exchange (FX) facility access for Enhanced Business Service (EBS) and non-EBS lines; PBX Direct Inward Dial and Direct Outward Dial (DID/DOD) interface by way of line trunks; and mobile/cellular type 1 interface by way of line trunks.

## Foreign Exchange (FX) facility access

FX facility access provides a single station user or an EBS group located outside a central office's local service area with access to services normally available only to local subscribers. A dedicated FX line trunk (a trunk that has line features and options assigned to it), used solely by the FX subscriber, provides this access; the FX subscriber pays a monthly mileage charge for the distance between the local serving office (*local station end*) and the distant serving FX office (*far CO end*).

Traditional FX service over T1 trunk facilities is supported through special service channel bank units and drop-insert units (see Figure 6-4). The drop-insert unit supports the splitting off of one of the 24 T1 channels. This allows the use of existing trunking facilities between central offices to transport the FX special service. With the Digital PX Trunks feature, the DMS-10 switch emulates the functions of an FXO, when the local station end functions as an FXS channel bank unit, and of an FXS, when the far CO end functions as an FXO channel bank unit (see Figure 6-5).

From the local station end, the subscriber accesses the far CO end either by way of an automatic routing feature, by dialing an access code, or by pressing an M5000-Series business set key which has been assigned automatic routing or the access code. The line trunk station at the local station end provides the connection to the far CO end. After the far CO end office has been accessed, dial tone is provided either by the local station end or the far CO end. When dial tone is provided by the local station end (*senderized* operation), the dialed digits are collected and outpulsed to the far CO end after signaling mode conversion (DP to DTMF, or vice versa), over the dedicated FX line trunk. The far CO end translates the digits and terminates the call, while providing all call progress tones. When dial tone is provided by the far CO end (*non-senderized* or *cut-through* operation), the digits (other than access digits) dialed by the subscriber are passed one-by-one over the dedicated FX line trunk as they are input and are translated by the far CO end office, which also provides all call progress tones.

Incoming FX calls to the local station end office from the far CO end office terminate to a station associated with the incoming FX line trunk. This trunk acts an incoming ringdown trunk (no digits are signaled) routed to a fixed station. The station may answer, forward, or perform any other service for the call.

**Figure 6-4: Traditional FX service**

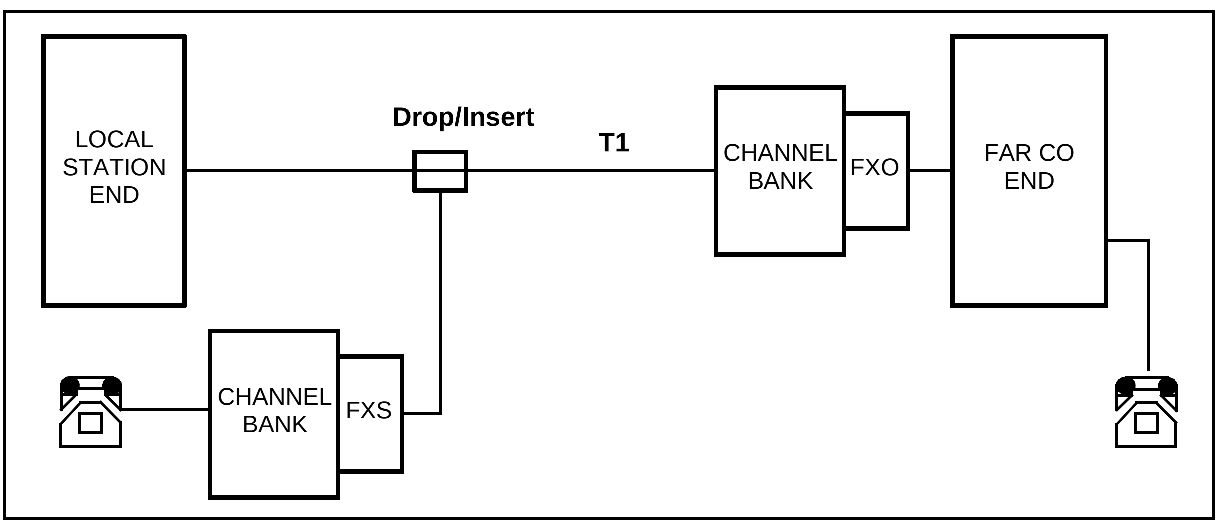
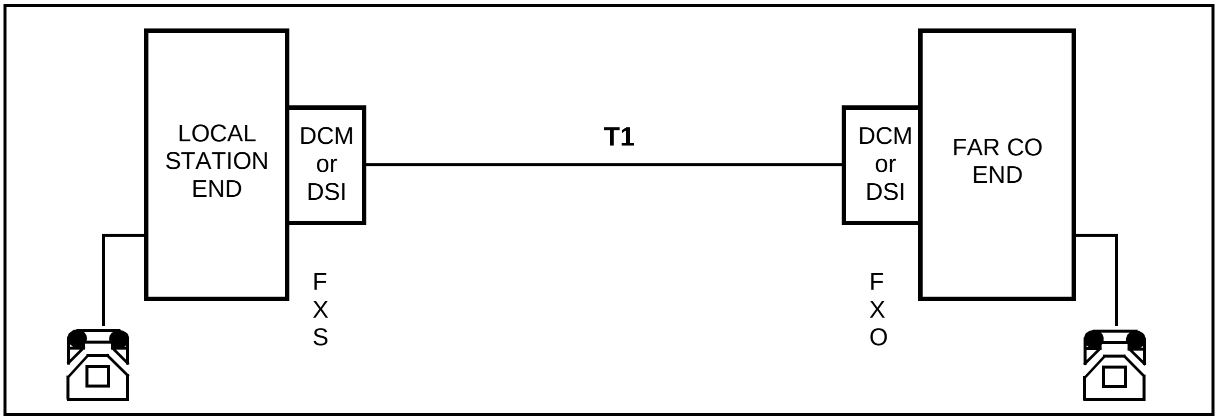


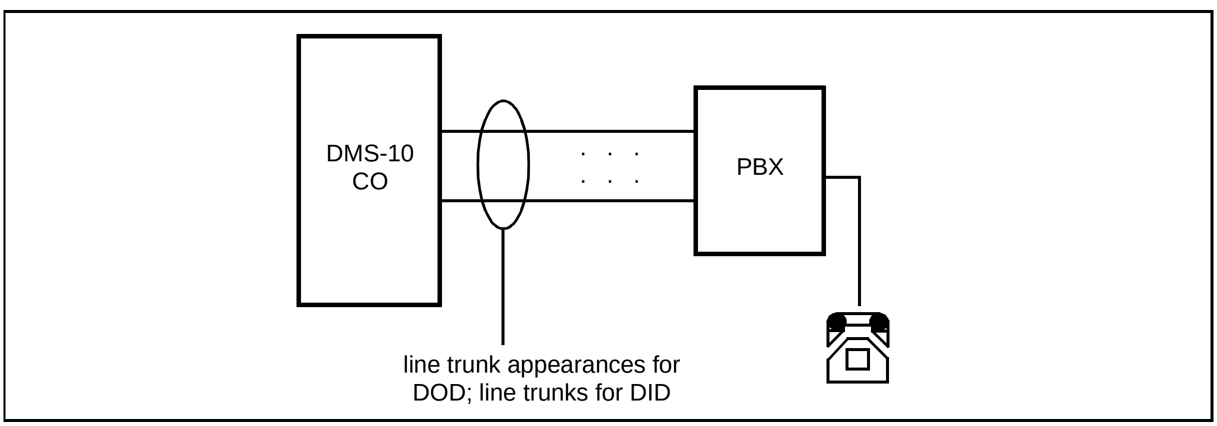
**Figure 6-5: FXO and FXS in a DMS-10 switch**




## PBX access

The Digital PX Trunks feature provides 1-way or 2-way Direct Inward Dial (DID) or Direct Outward Dial (DOD) interface to a PBX by way of a T1 carrier (see Figure 6-6). The DID capability enables a PBX station to receive calls from the exchange network without attendant assistance. After a 1-way outgoing trunk or 2-way line trunk connected to a PBX is seized, 1 through 7 digits (DP, MF, or DTMF) are outpulsed toward the PBX. The outpulsed digits uniquely identify the target station to the PBX. Answer supervision is received when the target station answers. The DOD capability enables a PBX station to place a call to the exchange network without attendant assistance. Normally, the PBX subscriber dials an access code and then dials up to 32 digits. After appropriate routing in the DMS-10 switch, answer supervision is returned to the PBX for local and toll calls.

**Figure 6-6: DID/DOD from PBX to CO**



## Mobile/cellular type 1 access

The mobile radio interface functions in the same manner as the PBX interface (see Figure 6-7 and the PBX access description above). The cellular type 1 interface allows origination and termination access to a cellular mobile carrier (CMC); only MF signaling is supported (see Figure 6-8).

## Conditions pertaining to the Digital PX Trunks feature

The following conditions apply to the Digital PX Trunks feature:

- FX line trunks are normally 2-way with ground start signaling mode; loop start is also supported.
- Tandeming of FX services is not supported.
- FX line trunks are considered non-unique directory numbers and will be identified as “out-of-area” to CLASS features.

**Figure 6-7: Mobile interface**

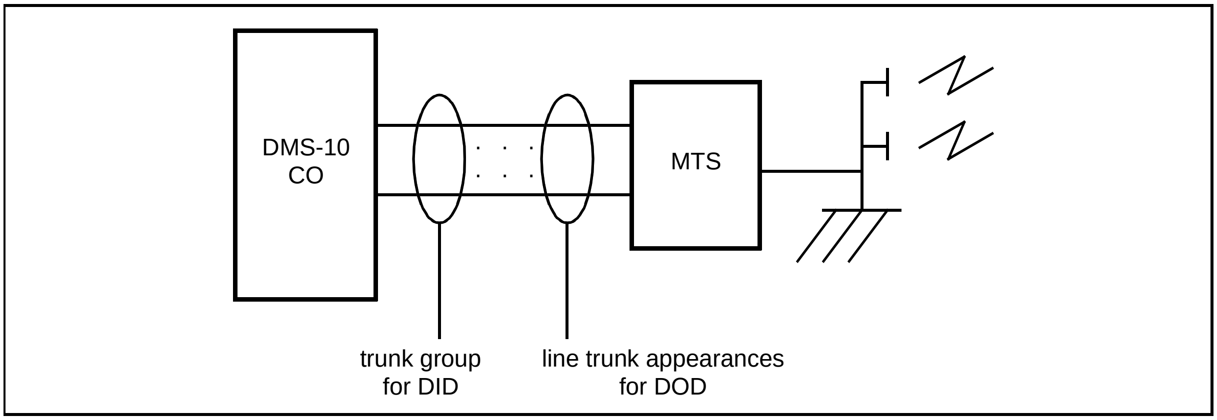
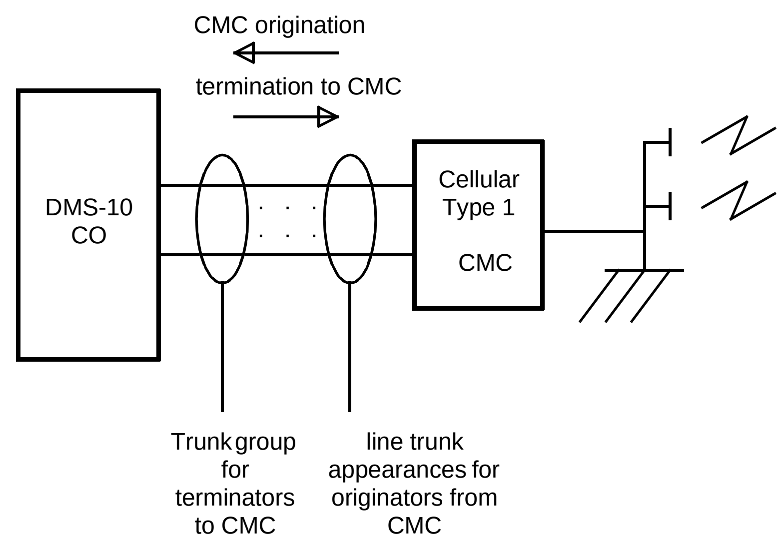


**Figure 6-8: Cellular Type 1 interface**




## Interaction of the Digital PX Trunks feature with other features

### Feature interaction with the FX feature

Features on the far CO end (FXO) line trunk station that require activation code dialing may be activated after the FX facility is reached by the subscriber. If the FX facility is accessed automatically upon going off hook, then subscribers cannot activate features assigned to their station in the local station end office because all digits dialed are processed by the far CO end office. If the FX facility is accessed by dialing an access code, subscribers have the option of activating features in either office.

The local station end (FXS) line trunk is not used to implement subscriber features but is used, instead, as a means of connecting the subscriber to the FXO station.

Because calls placed over the FX line trunk automatically terminate to the FX subscriber station, most origination type features, such as Presubscription, are not operable on an FXS line trunk station. Similarly, most terminating features, such as Call Waiting or Directory Number Hunting, are not operable on either the FXS or FXO line trunk stations since calls placed to these stations are routed over the next available FX line trunk and receive busy tone if the trunk cannot be seized. Call Forwarding can, however, be implemented on FXS or FXO stations to provide the subscriber with a means of transferring FX calls to a different number or facility.

#### Feature interaction with the PBX and cellular type 1 features

If the PBX has the capability of sending a switch-hook flash over the trunk to the DMS-10 switch, then that hookflash will be handled with respect to the features available on the line trunk station on which the call is established. Features requiring activation code dialing that are available on the PBX or cellular type 1 line trunk station can be activated by the subscriber on that station. Terminating features, such as Call Waiting and Directory Number Hunting, are not operable on a PBX or cellular type 1 station because calls terminating to these stations are routed directly over the associated trunk and receive busy tone or alternate routing treatment when an available channel cannot be seized. Call Forwarding, however, can be implemented to route incoming calls to specific stations, whether they are line trunks or normal lines.

### Digital PX Trunks feature billing

Billing for line trunks occurs on a station, rather than trunk group, basis. Unlike normal trunk originations, for which the TG number is stored in the AMA record, line trunk originations are treated like line originations for billing purposes and the DN of the line trunk is recorded in addition to the LTG number.

The called number stored in the billing register and AMA record for line trunk terminations are recorded differently depending on the type of line trunk: for FXS and FXO line trunks, the called number is the DN of the line trunk, but for PBX and cellular type 1 line trunk terminations, the called number is the dialed digits, as is done for normal outgoing trunk calls.

Although the FX subscriber's connection to the FXS line trunk is a free call and is not billed, Message Detail Recording can be used to track such FX calls. In these cases, billing registers are used to record the information.

## Voice Mail

The Voice Mail feature enables customers to maintain a voice messaging system (VMS) “mailbox.” Subscribers may use the mailbox either to leave messages for, or to receive messages from, callers who have access to their mailbox. The Voice Mail feature includes NTI's standard software interface to voice messaging systems, “Simplified message desk interface (SMDI),” and NTI's voice messaging system unit, “DMS-10 Voice Mail.”

### Simplified message desk interface (SMDI)

SMDI is a standard interface between the DMS-10 switch and a voice messaging system (VMS). SMDI is activated through a feature bit and applies to stations assigned the SMDI and Message Desk (MD) station options. Stations with the SMDI option connect to the VMS. Stations with the MD option are allowed to forward calls to the VMS and receive a message waiting indicator when a message has been left in the station's mailbox.

When a subscriber calls an SMDI station to leave a message, the DMS-10 switch first seizes a line to the VMS and then sends a corresponding message containing pertinent information about the call, including the calling party's number (if available), the type of call, the line on which the call terminated, and, for forwarded calls, the called party's number (which can be blocked from delivery) and type of call forwarding, to the VMS over a separate SMDI data link. After receiving this information, the VMS ensures that the call terminates at the correct mailbox and a recorded mailbox message is played.

After the call has terminated and a caller has left a message in the mailbox, the VMS sends a message to the DMS-10 switch over the SMDI data link to activate a message waiting indicator for the station associated with the mailbox as determined by the subscriber's station options (see Table 6-D). The subscriber retrieves the message by calling the VMS. When this occurs, the DMS-10 switch seizes a line to the VMS and sends the mailbox number (if available) to the VMS over the SMDI data link so that the message in the mailbox can be retrieved. After the subscriber has retrieved all of the messages, the VMS sends a message to the DMS-10 switch over the SMDI data link to deactivate the message waiting indicator for the subscriber's station. The subscriber may also deactivate the message waiting indicator by dialing a special code defined by the operating company.

To maximize utilization of lines to the VMS, the SMDI lines can be configured in hunt groups. A single hunt group can be configured to serve all subscribers or several hunt groups can be configured to serve separate business or subscriber groups. Instructions for setting up the SMDI feature can be found in Service Order Procedure (SOP) 0081 in the NTP entitled *Data Modification Manual* (297-3601-311).

## Simplified Message Desk Interface (SMDI) enhancements

Five enhancements have been made to SMDI: EBS Privacy Treatment, Message Waiting Indicator Lamp, Ten-digit Number Delivery to the VMS, Two-second Stutter Dial Tone, and Voice Mail Direct Control. Instructions for setting up the SMDI enhancements are included in the service order procedure (SOP) 0081 in the NTP entitled *Data Modification Manual* (297-3601-311).

### EBS privacy treatment

Delivery of directory numbers can be blocked for forwarded calls on an office-wide basis. This enhancement adds privacy treatment for individual EBS groups: delivery of calling numbers for forwarded intra-group calls can be suppressed for an entire EBS group.

### Message Waiting Indicator Lamp

This enhancement is activated through two new station options: Message Waiting Indicator Lamp (MWIL) and Suppress Intermittent Dial Tone (SIDT). The MWIL option causes a message waiting indicator lamp on a subscriber's customer premises equipment (CPE) to be turned on when message waiting activation occurs. The same type of message is sent to turn the lamp off when message waiting is deactivated. Intermittent (stutter) dial tone is still provided to stations with the MWIL option. If, however, a station doesn't require the intermittent dial tone, the SIDT option causes the tone to be suppressed when message waiting has been activated. Table 6-D shows the message waiting indications that are provided to a station based on the station's MD, MWIL, and SIDT station options.

### Ten-digit number delivery to the SMDI unit

This enhancement enables both 7- and 10-digit directory numbers to be delivered to the VMS unit for local and ISUP trunk calls.

### Two-second Stutter Dial Tone

This enhancement offers a choice between two forms of intermittent dial tone used for message waiting indication: continuous stutter dial tone or two-seconds of stutter dial tone followed by normal dial tone.

**Table 6-D:**
**Message Waiting indications for stations with MD, MWIL, and SIDT options**

| Station Options Assigned | Intermittent Dial Tone | Lamp |
|---|---|---|
| MD | YES | NO |
| MD and MWIL | YES | YES |
| MD, MWIL, and SIDT | NO | YES |
| MD and SIDT | NO | NO |

### Voice Mail Direct Control

This enhancement allows office-wide control over DN delivery for direct calls to the VMS when the DN of the calling party is available (that is, the call originates from a line or on an ISUP trunk) and the calling party's station is not in an EBS group that the message desk is also a member of. When the enhancement is active, the calling party's DN can be delivered to the VMS if office-wide message desk number blocking is not active (prompt MDNB = NO in the SYS (system) prompting sequence of overlay CNFG) and if the privacy status indicators that apply to the calling party's station so allow.

## Voice mail enhancements for ISDN

Like the non-ISDN voice mail, the SMDI data link is the standard interface between the DMS-10 switch and a VMS, and the SMDI feature bit must be activated. Only three directory number options, however, apply to ISDN voice mail. Those options, assigned through the DNCT prompting sequence, are; MD, NMD and SIDT. Functionally, the options are identical to the station options assigned to a non-ISDN station through the STN prompting sequence.

The MD option, assigned with a call forwarding feature, allows subscribers to forward calls to the VMS. If a feature indicator has been assigned in Overlay ISDN, prompting sequence TCGN, a message waiting visual indicator is received when a message has been left in the station's mailbox. The MD feature can only be assigned to a VI (voice information) DNCT. A VMS dialing sequence is also assigned with the MD feature.

A message waiting indicator (MWI) may be assigned using a Feature Activator and/or Feature Indicator, with a DN reference (FADN, FIDN, FFDN); otherwise the message desk feature functions in an identical manner to the conventional MBS message desk feature. The visual indicator provides un-retrieved message visual indication and the feature activator provides the ISDN subscriber with the ability to place a call to the voice mail system with a keystroke or similar action.

## Teen Service With Voice Mail

Teen Service with Voice Mail makes available separate VMS mailboxes for Teen Service subscribers' Primary Directory Number (PDN) and Secondary Directory Numbers (SDN). For complete information about this feature, see the Teen Service feature description in Section 4 of NTP 297-3601-105, *Features and Services Description*.

## SIP devices with Voice Mail

The station options MWIL and SIDT are not applicable for SIP devices. The only option required is MD. The DMS-10 has no control over how SIP devices notify its end users of a voice mail message. When a SIP subscriber has been left a voice mail, the DMS-10 will send a SIP protocol message indicating a voice mail system update. The actual visual or audible notification to the end user is determined by the SIP device.

## DMS-10 Voice Mail

The DMS-10 Voice Mail VMS consists of a floor-mounted cabinet containing up to four modular shelves, or “nodes.” The system is offered in 15 different configurations, from 8 through 64 ports (in four-port) increments. A DMS-10 Voice Mail system configured with 64 ports can support up to 5840 subscribers, with a total message storage capacity of 200 hours. Message storage can also be expanded by adjusting message disk size (300 or 600 Mb), the number of nodes, and the amount of message storage allocated to each subscriber. A single DMS-10 Voice Mail system can serve up to eight DMS-10 switches.

A 1200-baud, full duplex RS232 data link connects the DMS-10 Voice Mail and the DMS-10 switch and is used for call set up and MWI activation or deactivation. The only additional equipment required for the DMS-10 Voice Mail are a dedicated visual display unit (VDU) and printer used for system administration. Additional information can be found in the documentation supplied with the DMS-10 Voice Mail system.

## Interswitch Message Service

Interswitch Message Service is a collection of voice mail features and services that uses a Message Storage and Retrieval (MSR) system. The MSR system enables callers to leave messages and enables MSR client users to retrieve them. The MSR system also notifies client users when the status of a waiting message has changed, through Message Waiting Notification (MWN) features such as stutter dial tone or a visual indicator.

Client user stations may reside on the MSR system serving switch. If a client user station does reside on the MSR serving switch, internal messaging is used for notification about message status changes. If, however, the client user station is on a different switch from the MSR serving switch, TCAP messages are passed between the switches through the CCS7 network for notification about message status changes. Thus, operating companies are able to centralize message desk functions in their networks.

### Message Desk Service Interswitch (MDSI)

A voice mail system attempts to update a subscriber's Message Waiting Indicator (MWI ) data either when messages are left for the subscriber or when the subscriber retrieves messages. The Simplified Message Desk Interface (SMDI) feature enables the DMS-10 switch to update subscribers' MWIs in response to requests from voice mail systems connected directly to it, through SMDI ports. The Message Desk Service Interswitch (MDSI) feature allows the DMS-10 switch to also accept requests for MWI updates from voice mail systems connected to MSR serving switches, in the network. MWI update requests from MSR switches are sent to the DMS-10 switch in TCAP messages by way of the CCS7 network.

The sub-system number in the TCAP message sent to the DMS-10 switch identifies the message as an MDSI request. The message contains a destination number (DN of the subscriber), a voice message storage and retrieval (VMSR) identification number, and a calling directory number (DN associated with the MSR). The message may also contain bearer capability information. If the destination number is determined to belong to the DMS-10 switch and to also have the MD option assigned, processing of the request continues.

The DMS-10 switch then determines whether the request can be honored. Each subscriber indicates whether any MWI update request will be accepted or if MWI update requests from only specific MSRs will be accepted. If MWI update requests from only specific MSR systems will be accepted, screening will be performed between the subscriber's MSR information and the MSR information received in the TCAP message. This is performed by screening the VMSR identification number in the TCAP message.

Each MD subscriber is assigned an MSR index number that serves as the index to a table of MSR identification numbers and MSR DNs (DNs associated with the MSR) defined in the switch. This table, which is defined using the MSR prompting sequence in Overlay CNFG (see NTP 297-3601-311, *Data Modification Manual*) contains the MSR information that is used in screening.

If the MWI update request passes the validation and screening tests, the DMS-10 switch then updates the MWI data associated with the subscriber's DN. The message waiting indications then performed are shown in Table 6-D. After MWI updating is completed, the DMS-10 switch sends a TCAP response message that indicates success or failure of the update, back to the MSR that originated the request.

The following conditions pertain to the MDSI feature:

- MD subscribers on a DMS-10 remote unit in ESA mode are not able to have calls forwarded to the voice mail system or to call the voice mail system in order to retrieve calls.
- Office-wide parameters such as Message Desk Number Blocking (MDNB) or Voice Mail Direct Call do not apply to MDSI
- Subscribers to one of the call forwarding features should only be allowed to forward their phone to a voice mail system when they also subscribe to MD (for example, one of the TEEN options with a call forward mode of 2 or 4). Subscribers must be sure that the forward-to DN does not contain any special characters or access codes. The forward-to DN is screened against the MSR DN number which also does not contain special characters or access codes.

### Message Desk Serving Switch (MDSS)

The Message Desk Serving Switch feature enables the DMS-10 switch to act as an MSR serving switch. A DMS-10 MSR serving switch interfaces with the MSR system through SMDI links.

When calls forwarded from a client user on another switch terminate on a DMS-10 MSR serving switch, the DMS-10 MSR serving switch passes both calling party information and called party information on to the MSR system so that a message may be either left or retrieved.

When MWI data for a client user changes, an update request is sent by the MSR system to the DMS-10 MSR serving switch, over SMDI links. The switch then verifies the format of the message and attempts to identify the client user.

If the client user's station resides on the DMS-10 MSR serving switch, the client user's subscription parameters are examined. If the user subscribes to the MDSI feature, the MSRID of the SMDI link is used to perform screening. If the MSRID of the client user matches the MSRID of the SMDI link, MWI data for the user is then updated. If the user also subscribes to MWIL, internal messaging is used to update the message waiting indicator.

If an MWI update request is for a client user station that does not reside on the DMS-10 MSR serving switch, the switch sends a TCAP query containing the requested update function (activation/deactivation) to the client user's switch. The DMS-10 MSR serving switch then sets a timer and waits for a response. If the response is positive, the transaction is complete. If a response indicating an unsuccessful update attempt is received, or if the timer expires, the switch sends an error indication back to the MSR system over the SMDI links.

The following conditions pertain to the MDSS feature:

- MD subscribers on a DMS-10 remote unit in ESA mode are not able to have calls forwarded to the voice mail system or to call the voice mail system in order to retrieve calls.
- The MDSS feature requires that the SMDI feature and SMDI links must be configured in the DMS-10 MSR serving switch office.

## Local Data Base Services

Local Data Base Services (LDBS) are features designed for use with the LDBS platform. The features enable subscribers to block origination of ‘900' calls from their stations, substitute four-digit numbers for normal one- or two-digit speed dialing numbers, and block origination of calls with numbers beginning with the prefix ‘1.' Access to LDBS features is controlled by the feature bit LDBS in overlay CNFG (FEAT).

### Access codes

Subscriber access to individual LDBS features is controlled by a two-digit feature activation code and a subscriber Personal Identification Number (PIN). The feature activation codes are determined by the telco. When assigning the access codes, the telco must be careful to select codes that are not already reserved for use with other features. Subscriber PINs are determined by the subscriber. To change a PIN, the subscriber dials *DDAAAANNNN, where DD is the feature activation code, AAAA is the current PIN, and NNNN is the new PIN; within thirty seconds, the subscriber must then re-dial the same sequence in order for the changes to be made permanent.

## 900 Restriction

The 900 Restriction feature enables subscribers to block origination either of all 900 calls or specific 900 NXX calls from their stations. To activate or deactivate 900 call blocking, a subscriber dials a digit string in the form *DDAAAANNNC, where DD is the feature activation code, AAAA is the subscriber's Personal Identification Number (PIN), NNN is either the specific NXX to block or 000 (all NXXs are to be blocked), and C is either 2 (900 calls are to be blocked) or 8 (900 calls are to be allowed). Unsuccessful activation/deactivation calls are routed to a tone or announcement.

## Super Speed Calling

The Super Speed Calling feature enables subscribers to use four-digit numbers to retrieve their speed list numbers rather than the normal one or two digits. This enables mnemonics, such as ‘HOME,' to be used for speed calling. To add a speed calling number to the super speed calling list, a subscriber dials a digit string in the form *DDAAAANNNN, where DD is the feature activation code, AAAA is the subscriber's personal identification number (PIN), and NNNN is the speed calling number being added; within thirty seconds, the subscriber must then dial the number for which the speed calling number is being substituted in the form DD(1NPA)NXXXXXX (DD is the feature activation code). If a speed calling number is to be deleted, the subscriber dials *DDAAAANNNN, where DD is the feature activation code, AAAA is the subscriber's PIN, and NNNN is the speed calling number to be deleted; within thirty seconds, the subscriber must then dial DD0 (DD is the feature activation code) in order to make the change permanent. Unsuccessful activation/deactivation calls are routed to a tone or announcement.

## 1+ Blocking (bulk blocking)

The 1+ Blocking (bulk blocking) feature enables subscribers to block origination of calls with numbers beginning with the prefix ‘1.' To activate or deactivate 1+ call blocking, a subscriber dials a digit string in the form *DDAAAA000C, where DD is the feature activation code, AAAA is the subscriber's PIN, 000 are fill digits, and C is either 3 (block 1+ calls) or 7 (allow 1+ calls). Unsuccessful activation/deactivation calls are routed to a tone or announcement.

## 1+ Blocking (per call blocking)

The 1+ Blocking (per call blocking) feature requires subscribers to dial their access code before each 1+ call is placed. The subscriber dials a digit string in the form #AAAA. If an incorrect access code is dialed, the call is routed to a tone or announcement; otherwise, the subscriber receives second dial tone and can then dial the 1+ number.

## Time of Day for LDBS/AP

This LDBS feature enables the LDBS unit (DMS-10/AP platform) to query the DMS-10 switch for the current time, date, and day of the week. The feature supports routing of calls to an IEC based on day of the week, time of day, and the dialed NPA, NPA-NXX, or NPA-NXX-XXXX.

Standard Transaction Capabilities Application Part (TCAP) messages and a new subsystem, *LDBS Management*, are used to transport queries to and responses from the DMS-10 switch. A typical query and response sequence includes the following steps:

1) An initial query message is sent to the DMS-10 from the DMS-10/AP at intervals specified in the DMS/AP system. The request is routed to the DMS-10 directly or through an SRP.

2) The DMS-10 switch responds to the query with a response message that includes the time and date information.

3) If erroneous information is sent, the LDBS unit sends the DMS-10 switch an error message; the DMS-10 switch then outputs a message concerning this error message received from the DMS-10/AP.

### LDBS platforms

The following LDBS platforms provide the operating company with the capability of providing customers with Local Data Base Services (LDBS) features:

- LDBS 2000
- CNAM-DB
- Application Peripheral (J8T76A-1)

For complete information about each unit, refer to the documentation supplied by the manufacturer, Innovative Systems, LLC. For a description of the J8T76A-1 Application Peripheral, see Section 6, “Miscellaneous equipment,” in NTP 297-3601-150, *Equipment Identification*.

## Line Featured Trunk

Enables a special billing option and a flexible ANI digit to be assigned to analog and digital trunks.

## CFW Protocol Enhancements

The CFW Protocol Enhancements feature, also known as Internet Call Waiting, enables single-line internet users logged on to the internet to be notified of incoming calls. The notification is performed through a pop-up window on the internet user's PC screen that includes the name and number of the calling party.

The way in which the incoming call is processed is determined by the kind of service, either basic or extended, that the internet user subscribes to. Basic service subscribers receive incoming call notification only. Thus, default call handling is performed for the call, which may include either forwarding to voice mail or delivering busy tone. Extended service subscribers receive incoming call notification and then have 20 seconds in which to determine how the call is to be handled. The call handling options for extended service include taking the incoming call, forwarding the call to voice mail, forwarding the call to a pre-determined DN, or delivering a recorded message to the calling party. A greeting is played for the calling party while the selection is being made.

# Data Networking

Each 500-Series DMS-10 CPU core supports a pair of separate physical Ethernet connections. Both CPU cores share two IP addresses. This configuration enables simultaneous network access while in split mode and also continuous data transfer during a CPU switchover. Data networking allows remote software and generic upgrades through a DMS-10 Ethernet port.

## Ethernet

Ethernet is a multiple-access, packet-switched, LAN technology communications system for carrying digital data among locally distributed computer systems. The physical transport media used by the DMS-10 switch is twisted pair (10BaseT and 100BaseT). Ethernet uses a CSMA/CD (Carrier Sense Multiple Access/Collision Detect) transmission protocol. This protocol ensures that:

- all Data Terminal Equipment (DTE) on the network sense every transmission
- all DTEs have access to the network
- only one message is sent at a time

Although all DTEs sense the transmission, only the addressed DTE processes a message and sends back an acknowledgement. Ethernet allows a transfer rate of 10 Mbits/second. The IEEE 802.3u and IEEE 802.12 Ethernet standards support a transfer rate up to 100 Mbits/second. In an Ethernet system, when two DTEs simultaneously send signals, those signals collide. If this occurs, each DTE waits a random period of time before retransmitting the signal. For this reason, Ethernet best supports computer network environments with “bursty” packet-based communications.

Every computer attached to an Ethernet network is assigned a unique Ethernet address. Each packet contains the destination computer's Ethernet address. The packet reaches the destination Ethernet adapter which accepts the packet and passes it over to the operating system.

## Transmission Control Protocol/Internet Protocol

Transmission Control Protocol/Internet Protocol (TCP/IP) is a protocol set used for inter-networking. The entire set exceeds 100 protocols, of which TCP and IP are two. TCP is a transport layer protocol that provides connection-oriented end-to-end connectivities across networks. IP is software that keeps track of inter-network addresses for different nodes. In addition, IP routes outgoing messages and recognizes incoming messages. File Transfer Protocol (FTP) is another protocol in the TCP/IP suite of protocols. FTP provides access to files at remote sites linked to the Internet.

## IP Addresses

IP addresses are used to direct data traffic across a data network. An IP address is assigned to each network interface on all devices across TCP/IP-based networks. An IP address consists of four fields called octets. Each octet, separated by a dot, can be a number from 0 through 255. The four octets are divided into two identification addresses:

- network
- host (machine)

The number of octets in a network or host address depends on the network class, as does the maximum number of networks and hosts. There are three network classes (A, B, and C). Table 6-E shows the three network classes and their characteristics.

To better illustrate network classes, the following IP addresses are presented with the network address in bold and the host address in italics:

- Class A: **47**.*70.86.131*
- Class B: **144.125**.*33.201*
- Class C: **202.211.63**.*158*

Each octet contains 8 bits, therefore an IP address contains a total of 32 bits. To illustrate this concept, the binary equivalent of decimal IP address 144.125.33.201 is 10010000.01111101.00100001.11001001.

**Table 6-E: IP Addresses by Network Class**

| Class | Number of Octets | | Maximum Networks | Maximum Hosts |
|---|---|---|---|---|
| | Network | Machine | | |
| A | 1 | 3 | 126 | 16,777,214 |
| B | 2 | 2 | 16,384 | 65,534 |
| C | 3 | 1 | 2,097,152 | 254 |

## Subnetworks

Class C networks, which support the maximum number of networks, are often composed of subnetworks. These promote easier administration, greater security and increased network performance. A subnetwork (subnet) breaks large networks into multiple physical networks using separate smaller addresses. Typically, a subnet may represent all of the machines at one geographical location, in one building, or on the same local area network (LAN). Having an organization's network divided into subnets allows it to be connected to the Internet with a single shared network address. Subnets use subnet addressing to communicate through the network.

A Class B network allows 2 octets (16 bits) for the Network Address and 2 octets (16 bits) for the Host Address. Adding subnetting to this address requires that some portion of either the Network or Host address be used for the subnet address. Normally, subnet addressing uses the upper portion of the host part of the IP address to define subnet numbers. Table 6-F shows an example of this. The upper Host address octet of IP address 144.124.33.201 (33) is used for the subnet address.

**Table 6-F:**
**Subnet addressing**

| Network address | subnet address | Host address |
|---|---|---|
| 144.124 | | 33.201 |
| 144.124 | 33 | 201 |

## Subnet mask

A subnet mask indicates which bits are used to specify the network and subnet part of the IP address. Once a packet has arrived at an organization's gateway or connection point it can be routed through internal gateways using either the unique network address or the subnet number. The subnet mask instructs the router which bits to look at, and which to ignore. A mask is a screen of numbers. In a binary mask, “1" over a number indicates the number should be viewed and “0" indicates the number should be ignored. Using a mask saves the router from handling an entire 32 bit address.

Using the example in Table 6-F, the subnet mask should notify the router to look at the network address and the subnet address, but ignore the host address. The resulting subnet mask is: 255.255.255.0; with decimal 255 equivalent to binary 11111111.

## Routers

Routers are intelligent devices which connect multiple networks at the network layer of the protocol suite. Routers typically connect networks using the same high-level protocols. For example, a router would be able to connect an Ethernet network using TCP/IP to a Token Ring network also using TCP/IP. A router is able to determine the shortest path to a packet's destination. If one path fails, the router finds an alternate path to the destination host. A router is located at any juncture of networks or gateway.

A router creates or maintains a table of the available routes and their conditions and uses this information along with distance and cost algorithms to determine the best route for a given packet. Typically, a packet may travel through a number of network points with routers before arriving at its destination.

### Gateways

A gateway is a network point that acts as an entrance to another network. An internet network consists of gateway nodes and host nodes. Host nodes are user computers and computers that serve content (such as Web pages). Gateway nodes are computers that control traffic within a company's network or at a local Internet service provider (ISP). Gateways are devices that connect hosts or networks using different network protocol suites. Gateways control the flow of information in and out of networks, as well as perform data translation functions.

### Additional DMS-10 Data Networks Information

For additional information about the DMS-10 Data Networks feature, see NTP 297-3601-906, *DMS-10 Data Networks*.

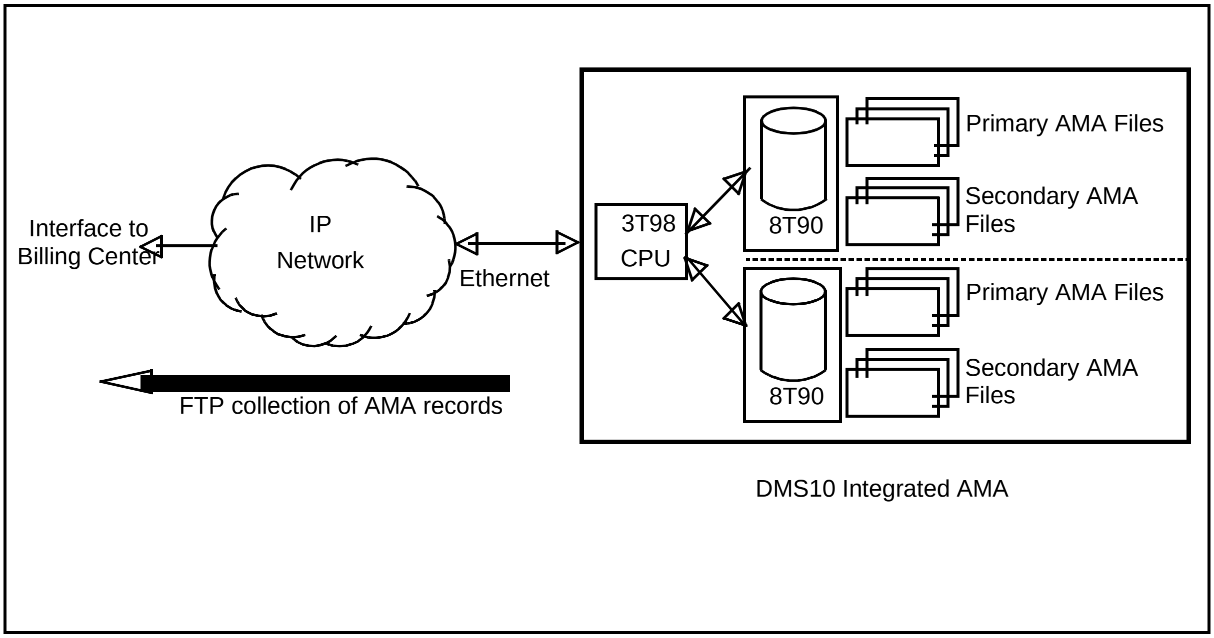
## Integrated Billing Storage and Retieval (IBSR) feature

Integrated Billing Storage and Retrieval (IBSR) is the DMS-10 feature that implements the Data Server component of the Telcordia AMADNS specification GR-1343-CORE. The IBSR feature will store AMA records securely on the new Redundant File System (RFS) that uses the native DMS-10 8T90 hard disk drives. The AMA records are formatted and arranged into Primary AMA files as specified by GR-1343-CORE. The Primary AMA files are then collected by the Data Processing and Management System (DPMS) via FTP. After collection, the Primary files are renamed as secondary files and remain stored on the DMS-10 Redundant File System until new Primary files overwrite them.

The IBSR feature is designed to eliminate the need for any external recording device, BMC Billing media converter, or 9 track tape.

The IBSR feature is applicable to either a stand-alone DMS-10 or for a cluster configuration of DMS-10 offices where the AMA collection is centralized at the Host Switching Office.

**Figure 6-9: IBSR feature overview**




## Trunk Group Member Usage (TGMU) feature

Trunk Group Member Usage (TGMU) feature is a feature for 505.10 that allows the incoming, outgoing, and two-way traffic on specified trunk groups to be studied. The TGMU feature will allow the DMS-10 to record, in a special file, all incoming and outgoing calls, answered and unanswered on the specified trunk groups. The TGMU data collected is preserved using AMA formats. The TGMU data records use a new Call Type Code of 951 with existing and new modules (module 900) appended to convey the call information. Analysis of the data is provided via vendor software packages.

The TGMU data records may only be collected when the DMS-10 is also configured for the Integrated Billing Storage and Retrieval (IBSR) feature. The IBSR feature is needed to segregate the TGMU data records from the AMA billing records normally generated. The TGMU data records are stored securely on the Redundant File System (RFS) that uses the native DMS-10 8T90 hard disk drives. The TGMU data records are formatted and arranged into primary TGMU data files located in the "misc/study/primary/tgmu" directory of the 8T90. The primary TGMU data files are then collected by a TGMU Data Collector via FTP. After collection, the primary files are renamed secondary files (located in the "misc/study/secondary/tgmu" directory) and remain stored on the DMS-10 Redundant File System until new primary files overwrite them.

The TGMU feature is applicable to either a stand-alone DMS-10 or for a cluster configuration of DMS-10 offices where the TGMU collection is centralized at the Host Switching Office.

## Call Logging (CLGS) feature

The Call Logging (CLGS) feature is a feature for 602.20 that allows subscribers to view call history on their lines by enabling an operating company to generate detailed call information about calls made to or from each Call Logging subscriber. The DMS-10 will gather and deliver the call information to a Subscriber Portal Server where the data will be formatted into a report for the Call Logging subscriber. Call Logging record delivery is provided in as close to real time as possible. Delays in processing occur by the necessary processing time in the DMS-10 after the call is completed. Call Logging information will be generated in addition to any normal AMA billing that may be collected for each call.

The format of the report to the Call Logging subscriber is dependent upon the service provider at the Subscriber Portal Server. This service is independent of the DMS-10 and is not covered in this document.

## Alarm Point Expansion

This feature for 504.10 increases the number of customer assignable alarm points on the DMS-10. An additional 64 alarm points are added. One of these alarm points is reserved for internal testing.

The interface to the remote alarm processor will utilize a 3T80BB circuit pack; therefore, no new DMS-10 hardware is required. The alarm scanning application (AlarmScanner) that manages and reports alarms received from the third-party Remote Alarm Processor is accomplished in SL-1 software.

AlarmScanner continuously polls the Remote Alarm Processor to gather alarms. Received alarm points data cause alarm messages to be sent to the software.

The Remote Alarm Processor resides in the MDF frame and communicates via RS-232 to the 3T80BB circuit pack. The communications protocol is TBOS (Telemetry Byte Orientated Serial), an industry standard. TBOS allows point-to-point communications between microprocessor-based monitored equipment and remote equipment through a serial link, minimizing the number of wires needed at each remote location.

Existing alarm point arrays are increased from 64 to 128 to accommodate the additional alarms. The expanded alarms will be customer-assignable using overlay ALRM. Alarm names are expanded from four characters to sixteen characters.

## Service Node Trunk Control

The Generic 504.10 Service Node Trunk Control (SNTC) feature consists of two complementary functions: Service Node Release (SNR) and Service Node Barge-In (SNB).

Service Node Trunk Control (SNTC) is implemented with high level (SL1) software changes and requires connections to a Service Node (SN), currently supported by the DMS-10. Service Node (SN) is defined as a peripheral device or another switch, capable of sending an ISUP Facility message. Service Node Release (SNR) provides the capability of 'dropping' the SN out of a call when its trunks and services are no longer required, eliminating the SN from the call path. Service Node Barge-In (SNB) provides the capability of 'adding' the SN into a call when requested.

The DMS-10 interfaces to the SN with Common Channel Signaling (CCS) links. The SN has its own point code, supports CCS#7 links, and utilizes ISDN User Part (ISUP) signaling, therefore simply looking like another switch to the DMS-10 (whether it is a switch or peripheral device).

The Service Node Release (SNR) function provides for the release of trunks, to and from the SN, when the trunks are no longer required for a call. The determination as to whether the trunks are still required is part of the SN and can be requested after the call is answered. When the SN determines that it (the SN) is no longer required in a call, it informs the DMS-10. This is accomplished by the sending of an ISUP Facility message (FAC) from the SN to the DMS-10. This message provides the Circuit Identification Code (CIC) of the two trunks connected to the SN. The DMS-10 attempts to merge the two calls into one, dropping the trunks to the SN. If successful, the DMS-10 simply releases the trunks to the SN. If unsuccessful, the DMS-10 does not release the trunks and no response is sent to the SN, therefore leaving the call in its current state.

The Service Node Barge-In (SNB) function enables the SN to insert itself into a call. To accomplish this, the SN sets up a call to itself through the DMS-10. It then informs the DMS-10 that it would like to barge into a call via an ISUP Facility Message (FAC) that includes the subscriber's DN to barge-in on. The DMS-10 finds the appropriate call and attempts to relink the call with new network paths to include the trunk connections from the SN. If successful, the previous call path is dropped and an ISUP FAC message is sent to the SN to indicate the success. If unsuccessful, an ISUP FAC message is sent to the SN to indicate the failure.

Billing does not change after a successful SNR or SNB. Each leg of the call is billed according to current requirements. If the incoming Initial Address Message (IAM) contains the charge number, the bill generated by the DMS-10 for the incoming leg of the original call and the call from the SN is against the received charge number. If the charge number is not provided and the calling party number is available, then the bill is made against the calling number. If neither the charge number nor calling party number is provided in the incoming IAM, the bill is generated against the incoming trunk. Once the barge-in has been completed and the new call paths established, charging continues as if the SNB feature was not encountered. The BRs in the call rings that were created before that barge-in are retained and moved to the new calls rings after the barge-in takes place.

## Dual Density ESMA

The Dual Density ESMA feature is an enhancement to the existing SCM-10A feature and introduces a new Multivendor Equipment bay. The Multivendor Dual Density equipment (MVDD) bay (NTQX90BA) provides the ability to configure 4 ESMA modules. These modules provide support for 96 P-side DS-1 links (24 per module). The increased number of ESMA modules enables a higher concentration of RDTs to be supported by a single bay of equipment.

There are no extension shelves in the MVDD bay and this limits the number of ISDN loops that can be supported. For this reason, the Dual Density ESMA is orientated towards POTS service.

## Online Output Message Manual feature

The DMS-10 currently communicates with human users mainly by displaying message numbers. Users must consult the current Output Message Manual (OMM) for the associated message definitions.

The Online Output Message Manual (O-OMM) will allow DMS-10 users to query the system for message number definitions. This will save time and eliminate the need for every Central Office (CO) and Network Operating Center (NOC) technician to have access to a CD or paper copy of the OMM.

The major features of the O-OMM include:

- A simple, highly available interface
- Context sensitivity
- Intelligent pattern matching
- Patchability

This feature provides the following customer benefits:

- Reduction or elimination of the cost associated with acquiring and distributing copies of the DMS-10 Output Message Manual every time the DMS-10 software is upgraded.
- Reduction or elimination of any cost associated with calls to DMS-10 Technical Support to determine the meaning of a message number that does not appear in the paper OMM (for example, a message added in a patch).
- Reduction in DMS-10 support costs due to faster and more efficient troubleshooting by technicians.
- Potential reduction in the cost of training personnel on DMS-10 support, due to the availability of online help.

- Faster fix turnaround for customer-discovered documentation problems. Currently, hard copies of the OMM are tied to DMS-10 software releases, so problems cannot be corrected until the next upgrade.

## Alarm Dispatch (ALDP)

This generic 504.10 feature allows for the administration of a call-out list for alarm reporting in the office. In the event of an alarm, the DMS-10 will call the first directory number (DN) in the list. The technician at that DN may or may not acknowledge the alarm. If the alarm is not acknowledged, the DMS-10 will call the next DN in the list, and so on, until the alarm is acknowledged.

Currently DMS-10 offers the Alarm Sending (ALSD) feature that provides alarm reporting to an operator at a remote location. When an alarm is detected, ALSD seizes a specific trunk and places a tone on it to indicate an alarm has been raised in the switch. ALSD can be configured to report catastrophic, major, and minor alarms in overlay CNFG (ALRM). ALSD will continue to report the alarm until it clears or until a switch technician calls the alarm checking directory number (DN) and acknowledges the alarm.

Alarm Dispatch (ALDP) is an enhancement to alarm sending. ALDP uses a call-out list of DNs to report the last raised alarm in the switch. Operators can add up to five DNs to the call-out list. These DNs can route to intra-office lines, intra-LATA/inter-LATA numbers, cell phones, pagers etc.

For every alarm raised, the system starts with the first DN in the list, and if a technician does not respond to the call, a call is made to the next DN in the call-out list. The technician responds to the call by calling the Alarm Checking (ALCK) number defined in the switch. A timer is started each time

ALDP makes a call, and it is when this timer expires, that ALDP will call the next DN in the call-out list. The timer gives the technician time to call the ALCK number to acknowledge the alarm and stop the alerting process.

ALDP can be set up to apply a tone or out-pulse a number when the called DN is answered. The tone indicates to the technician that an alarm has been raised in the office. If the call-out DN is to a pager, then the DMS-10 can leave the ALDP calling party number, indicating that an alarm has been raised. Once notified, the technician calls the ALCK number to determine the class of alarm. Note: The ALDP is compatible with pager systems that expect to receive only one sequence of digits upon answer.

> ***Note:*** *The ALDP feature is compatible with pager systems that expect to receive only one sequence of digits upon answer.*

# Facility Identification by Name

This 504.10 feature allows a descriptive alphanumeric name to be associated with selected facilities in the DMS-10. The facilities for which a name can be associated are:

- Standard TGs
- PRI (Primary Rate Interface) and non-PRI Line Trunk Groups (LTGs)
- PRI Simulated Facility Groups (SFGs)
- CALEA (Communications Assistance for Law Enforcement Agencies) Call Content Groups (CCGs)
- Routes
- Digital Signal Interface (DSI) Links (DSLKs).

Once the feature is enabled and facility names are assigned, the names are displayed along with the TG number, route number, and DSLK equipment location in the DMS-10 system output wherever possible.

Currently certain DMS-10 facilities are identified by a number or physical equipment location only. This feature provides the ability to define meaningful facility names, so that a craftsperson can more quickly identify the facilities that logs and other system outputs are concerned with.

## Feature operation overview

If the Facility Identification by Name feature is enabled in DMO CNFG, all functionality of the feature is activated; that is,

- Trunk group (TG) names may be assigned by the craftsperson in data modification overlays TG, PRI, and SURV
- Route names may be assigned in DMO ROUT
- DSI link (DSLK) names may be assigned in DMO NET
- Defined facility names are output whenever facility numbers/locations are output in the system, respectively.

No changes to system I/O are visible if the feature is disabled, other than the addition of a new prompt, PRFN (Print Facility Name), in DMO CNFG (SYS) to allow enabling the feature. The feature is initially disabled upon upgrade to generic 504.10.

Once the feature is enabled, all facility names are unassigned, and they remain so until set by the craftsperson. There is no requirement that any or all facilities be named after the feature is enabled.

If the feature is disabled after previously being enabled, any facility names that had previously been defined by the craftsperson are retained in data store, but all output of facility names is suppressed. If the feature is later re-enabled, the previously set facility names are again activated.

Facility names may consist of a mix of up to twenty-eight (28) alphabetic, numeric, and special characters, including any symbols in the printable ASCII range. Names need not be unique, and furthermore, can be unassigned. Blank spaces are also allowed within names.

Double quotation marks ("") are required when the facility name is input by the craftsperson at a DMO prompt. The quotation marks allow the DMS-10 software to process the name as a string that may include blank spaces and/or special characters. When the facility name is output, no quotation marks are printed.

## Administration

### Feature configurability

There is no feature bit for Facility Identification by Name; however, the feature is enabled and disabled via the new PRFN prompt in DMO CNFG, SYS (System) prompting sequence, as previously noted. Once the feature is enabled, all feature functionality becomes operational, and facility names can be assigned by the craftsperson in the appropriate overlay program.

A second prompt, FNOM (Facility Name Printing in Operational Measurements?), which is prompted only if the feature is enabled (PRFN=YES), allows the craftsperson to selectively disable facility name printing in Operational Measurements (OPMs) only.

### TG Name query

Queries to look up a TG/LTG/CCG number (or numbers) based on a TG name are provided in DMO ODQ (Office Data Query). Queries to look up unassigned TG/LTG/CCG names are also added. Since duplicate TG names are allowed, more than one TG number may be returned for a given TG name, and in this case, all associated TG numbers are listed.

### Other DMO Impacts

If the feature is enabled, all DMO programs that currently output a TG or route number on a query also output the name, if defined. The name is printed on the same output line as the number, in most cases. Similarly, all DMO programs that print a DSLK equipment location (CE b s p lk) now output the DSLK name, if defined, on the same line of output.

Note that for route names, if a GCON (Generic Condition) mnemonic is output, such as ?VCDN? (Vacant Directory Number), a route name will not be output in this case since the GCON names themselves serve as descriptive mnemonics. In the CNFG (GCON) prompting sequence, route numbers and any defined route names associated with each GCON can be queried.

### Operational measurements (OPMs)

If the feature is enabled, the TG name is output on a new line in several OPM blocks in addition to the TG number. However, OPM data sent to EADAS (Engineering and Administration Data Acquisition System) is not modified to include the TG name field, in order to avoid significant impact on downstream processing.

### Maintenance

#### Maintenance overlays

Several maintenance overlays output a TG number or DSLK equipment location (CE b s p lk) as a part of command status output. If the feature is enabled, any defined TG or DSLK names are also output on the TTY line, to the right of the TG number or DSLK location.

#### Maintenance messages

If the feature is enabled, all defined TG and route names are printed in DMS-10 system maintenance messages (logs) wherever a TG/route number is output today, in addition to the TG/route number. Similarly, defined DSLK names are printed in maintenance messages that currently output the DSLK location.

## Security Enhancement

The Generic 504.10 Security Enhancement feature makes the following enhancements to the DMS-10 system:

Password Consolidation: All passwords and accounts are only configurable by the administrative (root) user using the CNFG (ACCT)) prompting sequence in the NTP entitled Data Modification Manual (297-3601-311).

Password Integrity: Prior to 504.10 release, passwords are any combination of 4 or more characters, including sequences or series of characters which may be easily discovered. Passwords never expire, increasing the risk that a persistent hacker could infiltrate the system. To address these issues, DMS-10 administrators have the ability to enforce stricter security measures. Users may be required to use alphanumeric passwords, which could be subject to expiry after a configurable interval. The user account passwords themselves are encrypted before being written to disk, but a means is provided for Nortel support personnel to reset lost passwords upon customer request. Each DMS-10 TTY is equipped with an auto-logout capability that can be used to logout any user whose terminal has been idle for an extended period of time. Administrative users are able to enable and disable auto-logout on an individual TTY basis using the CNFG (LOGU) prompting sequence in the NTP entitled Data Modification Manual (297-3601-311).

Site Identification: Often, multiple DMS-10s are managed remotely by a central Network Operating Center (NOC). As a result, a technician may inadvertently open a telnet session to the wrong switch and make modifications to it before realizing the oversight. To safeguard against this occurrence, the site name is output before the DMS-10 TTY login prompt (!) and a resident "query site" command added.

User Identification: Logins to standard DMS-10 TTYs are virtually anonymous. Users may be aware of each other's presence, but no identifying information is available to distinguish between authorized and unauthorized personnel. By associating user name ids with TTYs, technicians can monitor access to the DMS-10 more closely and take appropriate action when an account has been compromised. A resident query user command can be used to determine the active users on the switch. TTYs requiring these separate user name logins can be specified by the administrative user using the CNFG prompting sequence in the NTP entitled Data Modification Manual (297-3601-311).

## Security Class of Service (SCOS)

This Generic 601.10 enhancement to the existing security system allows operating companies to define Security Classes of Service (SCOS) tables, each of which may specify different access levels to DMO and Maintenance overlays, as well as the ability to restrict or permit satellite-office access from a host office (when applicable).

The SCOS enhancement provides an additional level of security to augment the existing password-based security, but does not replace the existing password scheme (i.e., administrative class passwords will still be used and required, in addition to individual user passwords). Once SCOS usage is enabled, users will be required to login at most TTYs (excluding dedicated links) with an assigned user account name and password so that their overlay program access privileges can be determined. The assignment of SCOS tables to user accounts can be used to provide more restrictive access to overlay programs than is provided by the existing administrative password system when limiting user access to these programs is desired.

SCOS tables may also be assigned to restrict overlay access from physical (non-telnet) TTYs defined as dedicated links (DLINs), which do not require a login sequence. Telnet TTYs, on the other hand, will use the user name and password entered at telnet login time to determine the user’s SCOS privileges, if SCOS usage is enabled.

The SCOS privilege levels that may be specified for each DMO overlay program are:

- Full - allows modification of office data
- Query only - allows querying office data but no changes
- No access - disallows entering the overlay.

For each maintenance overlay, the SCOS privilege levels that may be specified are:

- Full - allows execution of all overlay commands
- No access - disallows entering the overlay.

Table 6-G provides an example of a simple SCOS table that might be set up by an operating company.

**Table 6-G:Example SCOS table**

| **SCOS 1** | |
|---|---|
| ACC SSO | YES |
| Full | DN, HUNT, DED, PED, CED |
| Query Only | CNFG, EQA |
| No access | All other overlays |

Up to thirty-one (31) SCOS tables may be defined by the operating company. SCOS 0 is pre-filled to provide full access privileges to all DMO and maintenance overlays (and use of the “ACC SSO” command if the office is a host in a cluster). SCOS 0 is the default setting until more restrictive SCOSs are defined and assigned to users and/or logical units. This SCOS table cannot be modified by the operating company. SCOS tables 1 through 31 are initially defined to allow no overlay access but may be redefined as desired.

The following SCOS-related data can be modified in overlay CNFG:

- SCOS usage on or off (PSWD prompting sequence)
- SCOS table assignment for user accounts (ACCT prompting sequence)
- SCOS table assignment for non-telnet dedicated-link TTYs (LOGU prompting sequence)
- SCOS table settings (SCOS prompting sequence).

For further information refer to the NTP entitled *Data Modification Manual* (297-3601-311P1).

It is envisioned that SCOS tables might be most likely utilized by operating companies to restrict access to a specified group of overlays for users who log into the system with the “ALL” password, or to further restrict users who log in with the “DMO” or “MTCE” password to a smaller subset of the overlay class or to query-only access within the DMO class. If Password Security is enforced (PWSC = YES in overlay CNFG-PSWD), the user will have access privileges only to those overlays allowed by his password class *and* those allowed by his assigned SCOS table. The more restrictive access level is always applied.

Logical units defined as dedicated links will continue their current operation of not requiring a login procedure; however, the SCOS table assigned to the TTY will determine overlay privilege levels for that TTY. The default access level will be full access (SCOS 0) so that no change will be apparent to the operating company unless a more restrictive SCOS is assigned to the TTY. If a user on a dedicated-link TTY enters the "LOGI" resident command and logs in with his user account and password, the SCOS table assigned to the TTY will take precedence over that assigned to the user.

The user's privilege level for a DMO overlay, whether full access or query-only, will be persistent throughout the course of the user's overlay session, even if changed by the root user while the user is active in that overlay.

When the user exits one overlay and requests another, the SCOS table associated with the user's account will be re-obtained from the user account file. At that point, any changes to either the SCOS table or to the SCOS assigned to the user account that were made by the root user during the previous overlay session will be applied.

## Feature Interactions

### I/O Scripting

If SCOS usage is enabled, the user's SCOS number (or the SCOS number assigned to the DLIN TTY) is obtained when an I/O script is requested by the "SCRP EXEC" resident command. This SCOS number is passed to the script TTY so that the user's SCOS privileges are applied to the executing script. This prevents a user from creating a script to perform commands that would be forbidden by his assigned SCOS table.

### Cluster

In a host office in a cluster with SCOS usage enabled, the "ACC SSO" command may be restricted by the SCOS table. If a host user's SCOS table grants privileges to execute this command, all subsequent overlay commands entered on the SSO virtual TTY are allowed. Similarly, in a satellite office with SCOS usage enabled, all overlay requests initiated from the host office will be granted.

## Restrictions and Limitations

SCOS restrictions are only applicable to manually loaded overlays; not those that may be scheduled to run in background or that may be initiated by the system automatically to correct a fault. SCOS restrictions are not applied to overlay SURV nor to debug/test overlays that require the DEBG administrative password to access.

SCOS settings are not used for Switching Control Center System (SCCS) dedicated TTYs. The operating company should ensure that SCCS TTYs are located in secure locations and that access to them is provided only to trusted users.

It is incumbent upon the operating company to set up password classes and assign SCOS tables to users in a logical manner; the DMS-10 software will not check for logical inconsistencies. For example, if a user were to use the DMO password to log into the system and password security were enabled, assigning an SCOS to his user account that permitted access to various maintenance overlays would not make sense and would not permit him access to these overlays, since the most restrictive security level is always applied.

## Network Archiving of Log Files

This feature is an enhancement of the I/O Logging feature introduced in Generic 503.10. I/O Logging captures all input to and output from every physical or telnet terminal connected to the switch. The data is stored on disk in a file that is created for each day. The user can then view and search the log via overlay LOG.

Currently, internal storage of logs is limited to a maximum disk space of 200 MB, and time-limited to the last 365 days. To increase capacity, the Network Archiving of Log Files feature provides a method to store and view the log files externally from the DMS-10.

The feature is implemented as part of the PASS program, which is freely available to Nortel customers via nortelnetworks.com. The feature is being released in conjunction with Generic 504.10, but can be used with 503.10.

## Input/Output (IO) Scripting

This Generic 504.10 Input/Output (IO) Scripting feature provides the ability to create and execute scripts on the DMS-10. These scripts issue a series of commands to the DMS-10 and may be initiated by a resident command or user-defined triggers which enable scripts to run at a specified time.

The DMS-10 overlay system updates necessary to support IO Scripting are concentrated in two main areas:

- the creation of a new overlay to implement a basic script editor and provide a trigger management system.
- the modification of overlay CNFG to enable the creation of a teletype (TTY) dedicated to script execution.

Redundant File System: The Redundant File System (RFS) is used to store User scripts and trigger information. This insures that the stored information is accessible as long as there is at least one DMS-10 hard drive in service. It also provides a mechanism that insures the ubiquity of information across both hard drives.

The UNIX file system (UFS) is used to store Vendor Scripts that will reside in a separate location. They are delivered as part of the generic or patch bundle process and as a result, access to them is disk dependent.

Script Logical Unit (LOGU): As scripts execute, the script receives output from the DMS-10 and commands generated by the script are sent to the DMS-10. This transfer of information between the script and the DMS-10 occurs on a newly defined Script LOGU. The CNFG (LOGU) prompting sequence will be updated to enable the configuration of this new type of logical unit. Script LOGUs are dedicated solely to script execution and are not available for use by any other user.

Overlay IOD: Updates are made to overlay IOD so that the DSBL TTY, ENBL TTY, and STAT TTY commands function correctly with Script TTYs.

Overlay SCRP: A new overlay is created to manage all User script and Trigger information. The user creates, changes, and deletes any User script or Trigger information. Both type of scripts (USER and VNDR) and triggers may be queried via the overlay. In addition, the user accesses Help information (if provided) for any script.

# Section 7: Maintenance features

## Introduction

The reliability and maintainability of the DMS-10 switch is enhanced by equipment redundancy, diagnostic programs, and alarm reporting features. Additionally, test lines and interfaces permit testing of subscriber loops and trunks, either from the DMS-10 site or from remote locations.

## Miscellaneous Maintenance Features

### PED Alarm Reporting

The PED Alarm Reporting feature audits lines and PE trunks for system-made-busy (SMB) status and, when a threshold for the maximum number of lines and PE trunks that may be SMB is met or exceeded, raises either a minor or major PED alarm. Operating company personnel determine the thresholds for both minor and major PED alarms on a site basis.

PED Alarm Reporting runs as part of background alarm auditing. The status of each PE, LCM, DMS-1U, and SLC-96 line and PE trunk associated with the office is examined. If a line or PE trunk is SMB, a running total of such lines and trunks for that site is incremented. After the audit of a site is complete, the total is compared with the thresholds set up by the craftsperson. If the total meets or exceeds the minor alarm threshold but is less than the major alarm threshold, a minor PED alarm is raised. If the total meets or exceeds the major alarm threshold, a major PED alarm is raised. If an alarm is raised at a remote site, it is also raised at the base site. The alarm may be cleared through the ALO or PED overlay. The alarm threshold is set in the SITE prompting sequence of Overlay CNFG (see NTP 297-3601-311, *Data Modification Manual*).

### Remote Power Alarm Delay

The Remote Power Alarm Delay feature extends the existing host switch capability of delaying a “no ac” (NAC) power indication for up to 15 minutes for RLCMs, RSLMs, RSLEs, OPSMs, and RSC-Ss connected to the host switch. The NAC can be delayed for 15 minutes only, for OPMs and OPACs connected to the host switch. The NAC alarm condition may be derived from different alarm points, depending upon the type of remote. The time delay is assigned using the ALRM (ALPT) overlay (prompt DELY) in NTP 297-3601-311, *Data Modification Manual*.

The following condition pertains to the Remote Power Alarm Delay feature:

- If an ac generator failure (ACG) alarm message occurs while an ac power fail message is being delayed, then the delay is aborted and the ac power fail alarm is raised immediately.

## Split Load

This feature provides a method by which a SYSLOAD is performed on the inactive CPU while call processing is maintained on the active CPU. The Split Load feature allows an operating company to upgrade an office from one generic to a higher generic with no interruption of service. This feature also provides specialized maintenance testing while the system is in the one-bus mode.

## Critical Memory Threshold Alerting (CMTA)

When the CMTA feature is activated, the amount of memory that is available for allocation as Data Store (DS) or Call Store (CS) is checked as part of scheduled memory audits or when the CHK MEM command in Overlay CED (see NTP 297-3601-506) is entered. When the available memory for either DS or CS is between 3000 and 1001 words, CMTA raises a minor DS/CS CNFG (memory configuration) alarm; when the available memory is 1000 words or less, CMTA raises a major DS/CS CNFG alarm. Operating company personnel clear the CNFG alarm by first adding or releasing enough memory to bring the total DS or CS memory above the CMTA threshold and then manually clearing the alarm (with the CHK MEM command). If adequate memory is added or released and the CHK MEM command is not entered, the DS/CS CNFG alarm will clear during the next scheduled memory audit. Until the alarm is cleared, an alarm message is output according to the schedule defined in Overlay CNFG (ALRM) (see NTP 297-3601-311) and also in response to the LIST ALM command in Overlay ALO (see NTP 297-3601-506).

The CMTA feature is activated by operating company personnel through the CMTA prompt in Overlay CNFG (CE).

## Custom Calling Data memory alarms

Custom calling features use Auxiliary Data Store (ADS) to store Custom Calling Data (CCD) entered by subscribers and operating company personnel. When CCD changes are made, additional ADS may be required or used ADS may be released. Custom Calling Data Memory Alarms help to ensure that a sufficient amount of ADS is always available. When it is determined, through system memory audits, ADS allocation or release, or the CHK MEM command in Overlay CED, that at least 80 percent of ADS has been exhausted, a minor CNFG (memory configuration) alarm is set; if at least 95 percent of ADS is exhausted, a major CNFG alarm is set. A major CNFG alarm is also set when a specific task requiring ADS cannot be performed because either adequate ADS is not available or ADS is not configured in the switch.

An ADS CNFG alarm is cleared either by releasing used ADS or by allocating additional ADS. A craftsperson allocates additional ADS through Overlay CNFG (CE) (see NTP 297-3601-311). When a craftsperson, or subscriber, performs a task that causes enough ADS to be released that the amount of ADS currently in use is equal to or below the release threshold defined for an existing alarm, that alarm will be cleared. Each release threshold is 10 percent below its respective alarm threshold: the release threshold for a minor alarm is 70 percent; the release threshold for major alarm is 85 percent. The release thresholds prevent alarms from being set, cleared, and reset when the amount of available ADS is just above an alarm threshold and tasks that use and release small amounts of memory, such as storing a directory number for call forwarding, are performed.

## Test lines and interfaces

The DMS-10 switch internally provides milliwatt, silent termination, and looparound test lines as well as a communication test line. The system also has internal facilities to test the subscriber's set.

In addition to the test facilities (that is, overlays and test trunks) and test equipment (that is, the ac tester, the Peripheral Maintenance System, and maintenance terminals) that are integral to the DMS-10 system, test interfaces are also provided for the following external equipment:

- No. 3 Test Cabinet (on-site)
- No. 14 Test Desk (remote)
- Loop Reporting System (LRS-1, LRS-10, LRS-100)
- Centralized Automatic Loop Reporting System (CALRS)
- Badger or Noller NP612 equipment
- Lordel T-9/15 Automatic Line Test Set (ALIT)
- Model 3703 Local Test Cabinet (LTC)
- Model 3704 Digital Remote Test Unit (DRTU)
- Model 1321 APC Subscriber Line Test Unit/ Subscriber Line Communications Unit (SLTU/SLCU)

For the SLC-96, channel and drop testing capabilities are provided through existing diagnostic overlays, test trunk, and test equipment interfaces in the DMS-10 switch. In addition to these test facilities available to the SLC-96, a No-Test-Trunk feature is provided as an interface for Mechanized Loop Testing (MLT) systems. Hence, automated testing is also available for SLC subscriber loops and channel unit testing.

For additional information about these testing facilities, see NTP 297-3601-500, *General Maintenance Information*.

## Integrated Trunk and Loop Testing (TLT)

Integrated Trunk and Loop Testing (TLT) is an optional feature that allows testing of trunks and subscriber lines for voltage, resistance, capacitance, and transmission (loss and noise). TLT uses the DMS-10 maintenance terminal to input test commands and to print the results.

A description of TLT and input information is contained in the NTP entitled *Maintenance Diagnostic Input Manual* (297-3601-506) under Overlay TLT.

## Test Line Loop Around Timing

Test Line Loop Around Timing is a feature designed to prevent non-traceable test line calls on the DMS-10 switch. This feature, implemented in Overlay ROUT, specifies the connection time allowed on a loop test call. Time limits can be from 0 to 255 seconds, at which time the connection is broken. Information on the overlay prompting sequences is contained in the NTP entitled *Data Modification Manual* (297-3601-311).

## Remote Office Test Line

Remote Office Test Line (ROTL) is a part of the Integrated Trunk and Loop Testing feature. ROTL serves as the software interface between the ac tester (ACT) in the DMS-10 switch and an external centralized trunk maintenance system, such as Centralized Automatic Reporting of Trunks (CAROT) or Automatic Transmission Measuring System (ATMS). ROTL can test trunks terminated by 100-, 102-, or 105-type test lines in the far-end office.

ROTL receives instructions concerning tests to perform and the identification of the trunks to perform the tests on through a sequence of priming digits sent by the external trunk maintenance system. Based on this information, ROTL uses the ac tester and the DMS-10 switching network to perform one of the following actions:

- originates a call to an outgoing trunk with an in-service status for transmission testing (loss, noise, gain/slope, noise with tone, manual)
- originates a call to an outgoing trunk with either a man-made busy, system-made busy, or lockout status for transmission testing (loss, noise, gain/slope, noise with tone, manual)
- performs a long-term (one-way loss and noise) and termination balance (measurement) test
- requests either the maintenance-busy status of an individual trunk or the number of man-made busy trunks in a given trunk group
- makes a security call back so that a trunk may be busied or restored to service by ROTL
- performs a connection appraisal test
- places a trunk being tested in made-busy state or returns the trunk to service

ROTL cannot test digital trunks on the Digital Synchronous Interface (DSI).

For more information concerning the ACT and the 100-, 102-, and 105-type test lines, refer to the NTP entitled *General Maintenance Information* (297-3601-500).

## 108 Test Line

The 108-type test line provides a non-inverting loopback for digital testing between two offices or between the central office and a private branch exchange (PBX). The loopback point is the DMS-10 network, on the same incoming or two-way trunk that receives the request for a loopback. The 108-type test line is also available for ISDN lines.

The originating office accesses the loopback using normal call processing procedures such as sending a seizure or outpulsing access digits. For ISUP trunks, normal call processing messages are exchanged. A telephone set or Datapath terminal can request the far-end local or PBX office for loopback on a digital and line trunk by normal dialing. An office of any class in the hierarchical switching network can also request a local or toll office in the network for a loopback by dialing from its test head connected to the trunks.

The office providing loopback activates the loopback if the incoming call is from a trunk or if the incoming digits match the office's loopback access code. When the test is in progress, regular call progress signals such as winks and acknowledgements are provided. The test can be aborted by either the originating or terminating office. If the terminating office aborts the test, the office removes the loopback before sending an onhook signal to the originating office and placing the trunk in an idle state.

The following conditions pertain to the 108 Test Line feature:

- When a BUSY command is issued using overlay DED (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*) for a Digital Carrier Module (DCM), Digital Signaling Interface (DSI), or DSI link, any loopback is dropped and the incoming trunks are indirectly disabled; a *return to service* (RTS) command is required to restore the trunk to service.
- During initialization and system software reload (SYSLOAD), calls to the 108-type test line are dropped.
- Network blocking that prevents network connection results in release treatment being applied to the trunk requesting the loopback.

## Synchronous test line

The synchronous test line is an incoming line that allows tests of the audible ringing, tripping, and supervisory functions of the DMS-10 office. Upon seizure of the test line, the following functions are performed:

1) Audible ringing followed by a 4-s silent period
2) Audible ringing for 2 s
3) Tripped ringing during a 3-s silent interval
4) Off-hook for 1.3 s followed by two 0.3-s off-hooks
5) Repeat of Step 4
6) On-hook

## Line test access

Line test access allows tests to be performed on DMS-1 subscriber lines by way of a metallic test pair terminating at the host DMS-10 switch. The test operator performs the tests from a test cabinet (for example, a No.14 LTD or a No.3 LTC). Both out-test to the DMS-1 subscriber loop and in-test to the line circuit are provided. Busy DMS-1 lines may also be monitored. For test data to be accurate, the test operator should compensate for the extra length in the metallic test wire.

The SLC-96 is configured with a metallic bypass pair that makes possible a full range of subscriber loop tests, including the channel unit test.

The RLCM is configured with either a metallic bypass pair or a DRTU to provide RLCM subscriber loop testing. The DRTU is used when the impedance between the DMS-10 switch and the subscriber telephone is greater than 1900 Ω.

The introduction of the RMM into the RLCM/OPM/OPAC provides operating company personnel with the ability to perform line circuit testing on subscriber lines at the remote site. The RMM allows all tests that are now performed on PE and LCE lines to be performed on RLCM lines. The Line Test Unit (NT2X10/NT2X11), Metallic Test Access pack (NT3X09), and the Incoming/Outgoing Test Trunk (NT2X90) are required to perform subscriber loop testing.

The Remote Carrier Urban (RCU) is configured with a metallic bypass pair that makes possible a full range of subscriber loop tests. An optional Line Test Access (LTA) circuit pack may also be provisioned in the RCU to enable external test systems such as a Mechanized Loop Tester (MLT) or Line Test Desk (LTD) to be used.

The DMS-10 switch can be configured (in Overlay CNFG (SITE)) to expect either 7 or 10 digits from incoming test trunks or Noller test trunks.

## Station ringer test

The station ringer test verifies that the ringer is working and that office data and outside plant assignment are correct. The Digitone pad test is also included. After all digits are dialed in sequence from the Digitone set, a return tone gives a pass or fail indication for each frequency. If a station also has the Calling Number Delivery (CND) or Usage-sensitive Calling Number Delivery (UCND) feature option, the test also causes the test pattern, “0123456789,” the current time of day, and the current date to be delivered to the subscriber's display unit. For a data unit associated with the Datapath Line Card feature, the test verifies correct operation of the dial pad, function keys, and indicator lamps.

## Dial speed test

The dial speed test is a method of testing the pulsing speed of the calling-station dial. When the directory number assigned for the test is dialed from the subscriber's set, a second dial tone is returned. The digit 0 is then dialed and one, two, or three bursts of high tone are returned, which indicate slow, normal, or fast dial speed respectively. The “dial speed normal” indicator is set for 8 through 12 pps.

## Stuck-coin test, first-trial failure

This feature facilitates preventative maintenance on coin stations by collecting and reporting information about stations that have stuck-coin test, first-trial failures. When this feature is configured, a stuck-coin test is automatically performed after either a coin collect or return function or after a call on coin overtime disconnects. If a stuck coin is detected during this test, the test is repeated a second time. If a stuck coin is detected during the second test, a maintenance message is printed on the maintenance terminal. If a stuck coin is not detected during the second test, the system pegs the CFTF (stuck-coin test, first-trial failure) operational-measurement (OPM) block; the CFTF OPM block printout shows the number of times a station has failed the first stuck-coin test but has passed the second stuck-coin test, along with the directory number and line-circuit identification number (physical location) of the station. The DMS-10 switch can report on up to 16 stations in the CFTF OPM block.

## Line Insulation Test (LIT)

Line Insulation Test, marketed as Automatic Line Insulation Testing (ALIT), is used to detect faults in subscriber loops. LIT is an overlay that uses the Peripheral Maintenance System (PMS) and the Line Test Unit (LTU) in the case of RLCM, to test for foreign battery, foreign ground, and tip-to-ring leak fault conditions. A foreign battery condition occurs when an absolute dc voltage or an ac voltage in excess of the default or user-specified value is measured on tip-to-ground or ring-to-ground. A foreign ground condition occurs when a resistance of less than the default or user-specified value is measured at tip-to-ground or ring-to-ground. A tip-to-ring leak occurs when a resistance of less than the default or user-specified value is measured at tip-to-ring.

Overlay LIT can be scheduled to run automatically through Overlay CNFG (LIT). The automatic testing is limited to 4 hours and is recommended during low-traffic periods (for example, 1 a.m. through 5 a.m.).

Subscriber loops that are call-processing busy are not tested. If a subscriber picks up the phone while the loop is being tested, dial tone will not be heard until the test is completed (within 4 seconds). Overlay LIT may be manually activated. In the manual mode, only one directory number at a time may be tested; in the automatic mode, the overlay tests as many loops as it can within the time the overlay is scheduled to run. When Overlay LIT is in the manual mode, operating company personnel can change the LIT parameters set in Overlay CNFG, query a directory number, and print a summary of the last cycle of test results. In the manual mode, resistance, capacitance, direct current, and alternating current tests for tip-ring, tip-ground, and ring-ground can be made in addition to the tests that are part of the automatic LIT testing.

Overlay LIT allows line insulation tests to be performed on the RLCM/OPM/OPAC lines. Overlay LIT uses the Line Test Unit and the Metallic Test Access card to perform voltage, resistance, and capacitance tests. Operating company personnel can perform these tests on individual line circuits or schedule the tests to be run at a given time.

Information on the Overlay CNFG prompting sequences are contained in the NTP entitled *Data Modification Manual* (297-3601-311); the input commands in maintenance Overlay LIT are contained in the NTP entitled *Maintenance Diagnostic Input Manual* (297-3601-506).

## LIT Predictor Interface

The LIT Predictor Interface allows LIT to be used in cluster applications. In this arrangement, each SSO outputs LIT data to the HSO or LCC using the Data Link Controller pack, and the data are transmitted to the Predictor Center in succession over a single serial data port provided at the HSO or LCC. More information about this feature can be found in the NTPs entitled *General Description* (297-3601-100) and *Input/Output System* (297-3601-300).

## LCE Line Card Monitor

This feature is designed to prevent NT6X17 and NT6X18 line cards from overheating due to damaging currents flowing into the cards from a subscriber's loop as the result of an abnormal condition. Normal line insulation testing is performed to determine whether a line's tip-to-ring, tip-to-ground, and ring-to-ground AC voltage, resistance, and DC voltage are within prescribed limits. If a line is determined to be “hazardous,” its cutoff (CO) relay is operated to isolate it from the outside plant.

For additional information about the operation of the LCE Line Card Monitor feature, see the NTP entitled *General Maintenance Information* (297-3601-500).

### Silent Switchman

This feature allows an outside plant craftsperson to test an open subscriber line without assistance from personnel in the Central Office (CO). Operating company personnel dial a particular access code (assignable by the operating company) from the subscriber's premises to open the metallic bus access relays to the subscriber's line and, after a ten-second busy tone, can perform a capacity test, a battery to ground test, and a tip and ring resistance test on the line. The length of time the line remains open is also assignable by the operating company, with the maximum time being 255 seconds. For a procedure used to set up this feature, see Service Order Procedure 0103 in NTP 297-3601-311, *Data Modification Manual*. For instructions about using this feature, see MP 1523 in NTP 297-3601-511, *Maintenance and Test Manual*.

### Dialable Cable Locator Tone

This feature is designed to assist the outside plant craftsperson locate a specific tip and ring cable pair without assistance from personnel in the central office. By dialing a particular access code (assignable by the operating company) followed by the DN of the desired line, the line associated with the DN is connected to the outgoing trunk, and an external tone source (required) is applied. The length of time for the tone source is 3-10 minutes, assignable by the operating company. When the pair is located, two hook flashes will stop the tone. For a procedure used to set up this feature, see Service Order Procedure 0098 in NTP 297-3601-311, entitled *Data Modification Manual*. For instructions about using this feature, see MP 1522 in NTP 297-3601-511, *Maintenance and Test Manual*.

## Electronic Software Delivery (ESD)

This feature provides operating company personnel with a web-based delivery mechanism for accessing new DMS-10 software generics and DMS-10 software patches to download. The Remote Generic Upgrade feature uses ESD for delivery of new generic loads from Nortel to a customer's DMS-10 site. The ESD feature consists of the following components:

- ESD system access
- Web-based client interface for DMS-10 software downloading and user/ product profile administration
- ESD application server
- Data transport
- DMS-10 software repository

## ESD system access

Only users registered through Nortel's eBusiness Common Registration and Secure Access Model may access the ESD system. The Nortel.Access Internet Link (NAIL) authentication mechanism provides users with access to designated system areas. NAIL provides a secure connection to information access both within and outside of Nortel's networks using Secure Socket Layer (SSL) encryption and decryption methods.

## Web-based client interface

The ESD system provides HTML links on the ESD web page for the following operations:

- DMS-10 generic upgrade/patch software listing and downloading
- user profile administration
- product profile administration

### DMS-10 generic upgrade/patch software listing and downloading

When this operation is selected, the ESD displays a web page with a list of available software for downloading. Operating company personnel may then search for particular files to be downloaded and may also select ranges of DMS-10 patches to package and download.

### User profile administration

The ESD maintains the following information that enables ESD operations to be performed:

- user ID
- user name
- user organization
- transport method
- IP address of user's workstation
- access profile

### Product profile administration

ESD provides information about available product releases for the DMS-10 switch. This information includes:

- product family
- software file name
- version number
- description of software

- file size
- information about software
- documentation
- date on which software was posted
- instructions
- bulletins
- related documentation

### ESD application server

The ESD application server performs the following functions:

- accepts requests from the web-based client
- interfaces with the user profile and product profile databases to validate users' requests for software downloading and listing
- interfaces with the DMS-10 repository back-end database to download software requested by a user
- sends responses to web-based clients containing either a list of the software to be downloaded or the actual software to be downloaded
- creates free disk space for future download requests
- records relevant details about downloads in the ESD database

### Data transport

ESD supports the following three methods of software delivery:

- drop box delivery - uses the existing network to place software loads in secure drop boxes for customer pick-up
- web distribution - provides operating company personnel access to Nortel software, using the internet
- network element - Nortel transfers software directly to the recipient's system.

The delivery method used depends on both product and user profile capabilities. Thus, only those transport methods that are supported are available. Operating company personnel are able to pause or stop downloads after they have been initiated.

### DMS-10 software repository

The DMS-10 software repository contains all of the DMS-10 software generic upgrade load files, patch files, and tool application software available to ESD users. ESD interfaces with the repository back-end in order to deliver the software requested by the ESD user.

# Move DS-30A Equipment (MDE)

The Move DS-30A Equipment feature enables operating company personnel to change peripheral loop assignments for in-service units connected either to a DS-30A pack (NT4T04) in a DMS-10 Classic Network or to a Network Interface pack (NT8T04) with DS-30A application in a DMS-10 expanded network configuration.

When an LCM is assigned in the DMS-10 switch, one, two, or three peripheral loop assignments on the DS-30A pack are made for the LCM. These peripheral loops must be on a single mated pair of DS-30A packs. Prior to the availability of the MDE feature, this assignment could not be changed without first de-populating the LCM (that is, removing all stations and line cards). The MDE feature enables operating company personnel to now change DS-30A loop assignments for an LCM without having to de-populate the LCM. Thus, the MDE feature enables DS-30A peripheral loop assignment changes to be made with minimal equipment outage time and with minimal effort.

A reassignment of DS-30A peripheral loops is made through Overlay UPDT (MNEW). Up to four “move” commands can be declared that specify equipment and loops to be reassigned. After the peripheral loops are physically re-cabled to their new locations, the reassignment is made active by performing a Sysload or Split Load.

The following conditions pertain to the MDE feature:

- LCMs at an RSC-S and remotes located off of an RSC-S are not supported by the MDE feature.
- The MDE feature does not impact current restrictions pertaining to peripheral loop assignments for LCMs and other equipment types. For example, peripheral loops assigned to an LCM must all be located on a single mated pair of DS-30A interface packs.
- Moving only part of the existing peripheral loop assignments for a unit is not allowed.
- System outage time during a move is dependent on time required for a system initialization (either Sysload or Split load).
- A maximum of four moves can be performed at one time in a DMS-10 Classic network configuration. A maximum of eight moves can be performed at one time in a DMS-10 expanded network configuration (DMS-10EN). A Sysload only supports up to four moves.
- The two links of a DSI functioning as a PRI interface must be connected to the same DS-30A.
- The DUMP command must be entered after any MOVE commands are entered, during the same session using Overlay UPDT; otherwise, the input data is lost.

# Section 8: VoIP Features

## Introduction

This section briefly describes the switched based Voice over Internet Protocol (VoIP) features offered by the DMS-10 switch. Note that additional features such as Three Way Call, Call Wait and User Transfer may be provided by the VoIP terminal.

The VoIP lines and packet trunks use the Session Initiation Protocol (SIP) call control signalling to interface to the DMS-10.

## Residential Line Features

The DMS-10 supports a VoIP line feature. The VoIP lines use the Session Initiation Protocol (SIP) call control signalling to interface to the DMS-10. The DMS-10 supports the following features on residential lines using VoIP terminals.

### Custom Calling Services (CCS)

#### Custom Calling Services access codes

Like non-VoIP subscribers, VoIP users wanting to use the Custom Calling Services supported by the DMS-10 will need to use the special codes required to activate and deactivate each service. Depending on the setup of a subscriber’s VoIP terminal, the operating company may need to alter the activation and deactivation codes for proper execution of the switch base services. For example, dialing an asterisk (*) from a VoIP terminal may have special meaning within the subscriber’s VoIP terminal. In this case the asterisk (*) may not be passed to the switch for translations, causing service access codes (SAC) in a format like *nn not to work. The operating company may need to define prefix translations such that services can be activated and deactivated without having to dial special characters like asterisk (*) and/or octothorp (#), such as dialing 11nn. For details about the translations, refer to overlay TRNS in the NTP entitled Data Modification Manual (297-3601-311).

#### Call Forwarding (CFW)

This feature is identical to the Custom Calling Services feature. For more information see "Call Forwarding (CFW)" on page 2-3.

#### Call Forward Busy (CFB)

This feature is identical to the Custom Calling Services feature. For more information see "User programmable Call Forward Busy Don't Answer (CFBD)" on page 2-6.

### Call Forward Don’t Answer (CFD)

This feature is identical to the Custom Calling Services feature. For more information see "User programmable Call Forward Busy Don't Answer (CFBD)" on page 2-6.

### Call Forward Fixed (CFF)

This feature is identical to the Custom Calling Services feature. For more information see "Fixed Destination Call Forwarding (CFF)" on page 2-4.

### Call Forward Remote Access (CFRA)

Subscribers using 2500-type terminals are required to have the digitone (DGT) option assigned to their stations in order to use the CFRA feature. Digitone is not compatible with VoIP lines. For a VoIP user to perform CFRA, the calling party’s VoIP device must be capable of generating DTMF tones. For more information see "Call Forward Remote Access (CFRA)" on page 2-7.

### Call Forwarding Limitation (CFL)

This feature is identical to the Custom Calling Services feature. For more information see "Call Forwarding Limitation (CFL)" on page 2-5.

### Speed Call (SSC and LSC)

This feature is identical to the Custom Calling Services feature. For more information see "Speed Calling (SSC and LSC)" on page 2-1.

## Custom Local Area Signaling Services (CLASS)

### Anonymous Caller Rejection (ACR)

This feature is identical to the residential CLASS feature. For more information see "Anonymous Call Rejection (ACR)" on page 2-17.

### Automatic Callback (ACB)

This feature is identical to the residential CLASS feature. For more information see "Automatic Callback (ACB)" on page 2-21.

### Automatic Recall (AR)

This feature is identical to the residential CLASS feature. For more information see "Automatic Recall (AR)" on page 2-24.

### Calling Identity Delivery and Suppression (CIDS)

The DMS-10 switch only allows a SAC to determine the delivery or the suppression of the subscriber’s number/name. If the call is marked as public or private in the originating SIP message, the switch will ignore this information and the call will continue with the subscriber’s default public/private status. For more information see "Calling Identity Delivery and Suppression (CIDS)" on page 2-29.

### Calling Name Delivery (CNAM)

For more information see "Calling Name Delivery (CNAM)" on page 2-27.

### Calling Name Delivery Blocking (CNAB)

The DMS-10 switch only supports a SAC to toggle the subscriber’s name to public. If the subscriber’s name is marked public in the originating SIP message, the switch will ignore this information and the call will continue with the subscriber’s name information marked private. For more information see "Calling Name Delivery Blocking (CNAB)" on page 2-28.

### Calling Number Delivery (CND)

For more information see "Calling Number Delivery (CND)" on page 2-35.

### Calling Number Delivery Blocking (CNB)

The DMS-10 switch only supports a SAC to toggle the subscriber’s number to public. If the subscriber’s number is marked public in the originating SIP message, the switch will ignore this information and the call will continue with the subscriber’s number marked private. For more information see "Calling Number Delivery Blocking (CNB)" on page 2-36.

### Calling Number Delivery Suppression (SUPR)

This feature is identical to the residential CLASS feature. For more information see "Calling Number Delivery Blocking (CNB)" on page 2-36.

### Customer Originated Trace (COT)

This feature is identical to the residential CLASS feature. For more information see "Customer Originated Trace (COT)" on page 2-36.

### Selective Call Acceptance (SCA)

This feature is identical to the residential CLASS feature. For more information see "Screening List Editing (SLE)" on page 2-37.

### Selective Call Forwarding (SCF)

This feature is identical to the residential CLASS feature. For more information see "Screening List Editing (SLE)" on page 2-37.

### Selective Call Rejection (SCR)

This feature is identical to the residential CLASS feature. For more information see "Screening List Editing (SLE)" on page 2-37.

### Simultaneous Ringing (SRNG)

This feature is identical to the residential CLASS feature. For more information see "Simultaneous Ringing (SRNG)" on page 2-38.

### Telemarketer Call Screening (TELE)

This feature is identical to the residential CLASS feature. For more information see "Telemarketer Call Screening" on page 2-42.

## Voice over IP (VoIP) Services

The following services are available to VoIP subscribers.

### Call Forward Internet Down (CFID)

The Call Forward on Internet Down (CFID) feature may be assigned as a station option to VoIP gateway lines. When a subscriber has either lost service, such as during a commercial power outage or when there is an outage with their broadband service provider, has not registered their Session Initiation Protocol (SIP) device, or when no response is received from the subscriber's VoIP gateway line, then calls presented to their Directory Number (DN) can not terminate; the caller will either receive ring-back tone or a release treatment will be provided. With the CFID feature assigned and activated, then these calls will be forwarded; for example, to the subscriber's voice mail or to a legacy line on the Time Division Multiplex (TDM) network.

### Call Reference Busy Limit (CRBL)

Call Reference Busy Limit, CRBL n(n), where n(n) is a value from 1 to 12; default 2. Indicates the total number of simultaneous call appearances the DMS-10 allows to be active for a VoIP terminal when interacting with the features ACB, AR and RAG. This number will vary based an the VoIP terminal product being assigned to the customer. CRBL does not apply if ACB, AR and RAG are not involved in the call.

## Voice Mail (MD)

To provide voice mail services for VoIP subscribers, the Message Desk (MD) option is required. The station options MWIL and SIDT are not applicable. The DMS-10 has no control over how VoIP devices notify its end users of a voice mail message. When a VoIP subscriber has been left a voice mail, the DMS-10 will send a message indicating a voice mail system update. The actual visual and/or audible notification to the end user is determined by the VoIP device. For more information see "Voice Mail" on page 6-43.

# Centrex IP Features

The DMS-10 supports a VoIP line feature. The VoIP lines use the Session Initiation Protocol (SIP) call control signalling to interface to the DMS-10. The DMS-10 supports the following features on Centrex lines using VoIP terminals.

### Centrex access codes and dialing plan

Like non-VoIP subscribers, VoIP users wanting to use the Centrex Services supported by the DMS-10 will need to use the special codes required to activate and deactivate each service. Depending on the setup of a subscriber’s VoIP terminal, the operating company may need to alter the activation and deactivation codes for proper execution of the switch base services. For example, dialing an asterisk (*) from a VoIP terminal may have special meaning within the subscriber’s VoIP terminal. In this case the asterisk (*) may not be passed to the switch for translations, causing service access codes (SAC) in a format like *nn not to work. The operating company may need to define prefix translations such that services can be activated and deactivated without having to dial special characters like asterisk (*) and/or octothorp (#), such as dialing 11nn. For details about the translations, refer to overlay TRNS in the NTP entitled Data Modification Manual (297-3601-311).

Some VoIP terminals may not support the Centrex dialing plan, or a digit map change on the device may be required to allow the Centrex dialing patterns.

## Integrated Business Services (IBS)

IBS allows a business to integrate up to six lines into a single customer group. The DMS-10 switch can support up to 255 IBS groups. Centrex IP provides IBS group and station features to VoIP subscribers. VoIP terminals and 2500 sets can be mixed within a group.

Subscribers using 2500-type terminals are required to have the digitone (DGT) option assigned to their stations in order to have certain IBS features. Digitone is not compatible with VoIP lines.

The following IBS features can be applied to VoIP lines.

### Intercom (INT)

The intercom (INT) feature allows an IBS user to call other stations within the same IBS group by dialing an octothorp (#) followed by the INT code of the called station.

Some VoIP terminals may not support Intercom dialing, or a digit map change on the device may be required to allow the Intercom dialing pattern.

For more information see "Intercom (INT)" on page 3-2.

### Call Hold (CHD)

This feature is identical to the IBS CHD feature. For more information see "Call Hold (CHD)" on page 3-2.

### Call Pickup (CPU)

This feature is identical to the IBS CPU feature. For more information see "Call Pickup (CPU)" on page 3-3.

### Directed Call Pickup Without Barge-in (DCPU)

This feature is identical to the IBS DCPU feature. For more information see "Directed Call Pickup Without Barge-in (DCPU)" on page 3-4.

### Directed call pickup with barge-in (DCBI)

This feature is identical to the IBS DCBI feature. For more information see "Directed call pickup with barge-in (DCBI)" on page 3-6.

### Directed Call Pickup Barge-in Exempt (DCBX)

This feature is identical to the IBS DCBX feature. For more information see "Directed Call Pickup Barge-in Exempt (DCBX)" on page 3-7.

### Directed Call Pickup Exempt (DCPX)

This feature is identical to the IBS DCPX feature. For more information see "Directed Call Pickup Exempt (DCPX)" on page 3-7.

### Directed Call Pickup from Any Station (DPUA)

This feature is identical to the IBS DPUA feature. For more information see "Directed Call Pickup from Any Station (DPUA)" on page 3-7.

### Busy Transfer (BTF, BTFA, BTFI)

The suite of busy transfer features (BTF - busy transfer, BTFA - busy transfer all, and BTFI - Busy transfer intragroup) allows calls to a busy station in the IBS group to be rerouted automatically to another station (the target DN) within the group.

Since SIP lines use local client-based call waiting, incoming calls to a SIP line with the BTF feature assigned will continue to be presented to the SIP line until the client device returns a ’486 Busy Here’ indication. For example, if the SIP client supports two call appearances and client-based call waiting is enabled, BTF features will not operate until the client has one call active and one call waiting. If the client supports more than two call appearances through additional line keys, BTF features will not operate until all call appearances are in use and the SIP client returns a busy indication to the next incoming call.

For more information see "Busy Transfer (BTF)" on page 3-8.

### Call Transfer Outside (CTO)

The CTO feature enables all IBS subscribers with user transfer capability to transfer an established call to two subscribers outside of the IBS group by performing a switch-hook flash, dialing the third party, and performing a disconnect at any time following completion of dialing.

All SIP Gateway lines in IBS groups have CTO capability, using the SIP client-based call transfer capability. The DMS-10 UTF option is not compatible with SIP lines.

For more information see "Call Transfer Outside (CTO)" on page 3-10.

### Don't Answer Transfer (DAT)

This feature is identical to the IBS DAT feature. For more information see "Don't Answer Transfer (DAT)" on page 3-10.

### Call Forwarding (CFW)

The IBS CFW feature is identical to the residential Custom Calling Services feature. For more information see "Call Forwarding (CFW)" on page 8-1.

### Call Forwarding Limitation (CFL)

The IBS CFL feature is identical to the Custom Calling Services feature. For more information see "Call Forwarding Limitation (CFL)" on page 2-5.

### Call Forward Busy Don't Answer

The IBS Call Forward Busy feature is identical to the Custom Calling Services feature. For more information see "User programmable Call Forward Busy Don't Answer (CFBD)" on page 2-6.

### Ring Again (RAG)

The IBS RAG feature is identical to the Custom Calling Services feature.

### Ring Again Denied (RAGD)

The IBS RAGD feature is identical to the Custom Calling Services feature.

### Convenience Dialing (CVD) and Convenience Dialing Controller (CVDC)

These features are identical to the IBS CVD and CVDC features. For more information see "Convenience Dialing (CVD) and Convenience Dialing Controller (CVDC)" on page 3-18.

### CLASS on Centrex

The IBS CLASS features are identical to the residential CLASS features. For more information see "Custom Local Area Signaling Services (CLASS)" on page 8-2.

### Call Forward Internet Down (CFID)

The IBS CFID feature is identical to the residential CFID feature. For more information see "Call Forward Internet Down (CFID)" on page 8-4.

### Call Reference Busy Limit (CRBL)

The IBS CRBL feature is identical to the residential CRBL feature. For more information see "Call Reference Busy Limit (CRBL)" on page 8-4.

## Enhanced Business Services (EBS)

The EBS feature package offers custom calling type features to multiline business subscribers. EBS allows the business customer to integrate separate access lines into a single communication group without special premises equipment.

A business subscribing to EBS is designated as an EBS customer group. A DMS-10 switch equipped with EBS can support up to 512 groups and up to 4,000 members per group. Centrex IP provides EBS group and station features to VoIP subscribers. VoIP terminals, 2500 sets and Meridian Business Sets (MBS) can be mixed within a group.

Subscribers using 2500-type terminals are required to have the digitone (DGT) option assigned to their stations in order to have certain EBS features. The digitone option is not compatible with VoIP lines.

Some SIP client devices may not support one-digit Station-To-Station (STS) dialing; two- to five-digit STS dialing patterns are recommended for any EBS group including SIP lines. Digit map changes on the device may be required to allow the STS dialing.

The following EBS features can be applied to VoIP lines.

### Call Pickup Group (CPUG)

This feature is identical to the EBS CPUG feature. For more information see "Call Pickup Group (CPUG)" on page 3-21.

### Directed Call Pickup Without Barge-In (DCPU)

This feature is identical to the EBS DCPU feature. For more information see "Directed Call Pickup Without Barge-In (DCPU)" on page 3-22.

### Directed Call Pickup With Barge-In (DCBI)

This feature is identical to the EBS DCBI feature. For more information see "Directed Call Pickup With Barge-In (DCBI)" on page 3-23.

### Directed Call Pickup Barge-In Exempt (DCBX)

This feature is identical to the EBS DCBX feature. For more information see "Directed Call Pickup Barge-In Exempt (DCBX)" on page 3-23.

### Directed Call Pickup Exempt (DCPX)

This feature is identical to the EBS DCPX feature. For more information see "Directed Call Pickup Exempt (DCPX)" on page 3-23.

### Directed Call Pickup From Any Station (DPUA)

This feature is identical to the EBS DPUA feature. For more information see "Directed Call Pickup From Any Station (DPUA)" on page 3-24.

### Don't Answer Transfer (DAT)

This feature is identical to the EBS DAT feature. For more information see "Don't Answer Transfer (DAT)" on page 3-24.

### Speed Calling (SSC and LSC)

These features are identical to the EBS SSC and LSC features. For more information see "Speed Calling (SSC and LSC)" on page 3-24.

### Group Speed Calling and Group Speed Calling Controller (GSC and GSCC)

These features are identical to the EBS GSC and GSCC features. For more information see "Group Speed Calling and Group Speed Calling Controller (GSC and GSCC)" on page 3-25.

### Call Hold (CHD)

For more information see "Call Hold (CHD)" on page 3-26.

### Busy Transfer (BTF, BTFA, BTFI)

The suite of busy transfer features (BTF - busy transfer, BTFA - busy transfer all, and BTFI - Busy transfer intragroup) allows calls to a busy station in the EBS group to be rerouted automatically to another station (the target DN) within the group.

Since SIP lines use local client-based call waiting, incoming calls to a SIP line with the BTF feature assigned will continue to be presented to the SIP line until the client device returns a ’486 Busy Here’ indication. For example, if the SIP client supports two call appearances and client-based call waiting is enabled, BTF features will not operate until the client has one call active and one call waiting. If the client supports more than two call appearances through additional line keys, BTF features will not operate until all call appearances are in use and the SIP client returns a busy indication to the next incoming call.

For more information see "Busy Transfer (BTF)" on page 3-26.

### Call Transfer Outside (CTO)

The CTO feature, an enhancement to user transfer, is assigned on a per-EBS group basis. CTO enables an EBS user to transfer an established call to two subscribers outside of the EBS group.

For SIP lines, the call transfer is performed using instructions for the particular SIP client device. All SIP Gateway lines in EBS groups have CTO capability, using the SIP client-based call transfer capability. The DMS-10 UTF option is not compatible with SIFor more information see "Call Transfer Outside (CTO)" on page 3-27.

### Call Forwarding (CFW)

This feature is identical to the EBS CFW feature. For more information see "Call Forwarding (CFW)" on page 3-27.

### Call Forwarding Limitation (CFL)

This feature is identical to the EBS CFL feature. For more information see "Call Forwarding Limitation" on page 3-28.

### User Programmable Call Forward Busy Don't Answer

These features are identical to the EBS CFB and CFDA features. For more information see "User Programmable Call Forward Busy Don't Answer" on page 3-28.

### Call Waiting Origination (CWTO)

The CWTO feature allows an EBS subscriber to impose call waiting on another subscriber who is in the same EBS group and does not have the CWT option assigned.

A SIP Gateway line can be assigned the CWTO option and may impose call waiting on another (non-SIP) member of the EBS group. However, call waiting cannot be imposed on a SIP line by another member of the group since the SIP call-waiting function is controlled locally by the SIP client device.

For more information see "Call Waiting Origination (CWTO)" on page 3-29.

### Dial Call Waiting (DCWT)

The DCWT feature allows an EBS subscriber to impose call waiting on another subscriber who is in the same EBS group and does not have the CWT option assigned. DCWT is the same as the Call Waiting Origination feature except that the originator must use an access code to impose call waiting.

A SIP Gateway line can be assigned the DCWT option and may impose call waiting on another (non-SIP) member of the EBS group. However, CWT cannot be imposed on a SIP line by another member of the group since the SIP call-waiting function is controlled locally by the SIP client device.

For more information see "Dial Call Waiting (DCWT)" on page 3-30.

### Ring Again (RAG)

The EBS RAG feature is identical to the Custom Calling Services feature.

### Ring Again Denied (RAGD)

The EBS RAGD feature is identical to the Custom Calling Services feature.

### Restricted Station Options (RES1, RES2, LOCO)

These features are identical to the EBS RES1, RES2, and LOCO features. For more information see "Restricted Station Options (RES1, RES2, LOCO)" on page 3-32.

### Call Park

Call Park enables members of an EBS group to park a call against their DN or Group Intercom (GIC) member number and continue to originate and receive calls on that set or MBS key. The Call Park feature is available in two versions: *Call Park* and *Directed Call Park*. Either 500/2500 telephone sets, M5000-Series business telephone sets or IP terminals may be used with this feature.

This feature is identical to the EBS Call Park feature. For more information see "Call Park" on page 3-32.

### Camp-On

The Camp-On feature enables a member of an EBS group to use the Call Transfer feature to extend a call originating either from inside or outside of the group to a busy station. Either 500/2500 telephone sets, M5000-Series business telephone sets or VoIP SIP terminals may be used with this feature. For more information see "Camp-On" on page 3-35.

The behavior of a SIP Camp On administrator differs from that of a TDM Camp On administrator (using a 500/2500 set or an M5000-Series business set) in the manner in which the calling party is treated. Once the SIP administrator is provided confirmation tone, the call is dropped; fast busy tone will be heard. The SIP administrator will not be able to reconnect to the calling party or cancel the Camp On request.

A SIP device may also be the target of a Camp On request when a ’486 Busy Here’ indication is returned when attempting to terminate to the SIP device and only one other active call exists (i.e., SIP client does not have call waiting). The SIP Camp On target is not provided Call Waiting tones. Once the SIP devices disconnects, it is alerted immediately indicating there was a waiting call.

### Virtual Facilities Group Controls (VFGC)

This feature is identical to the EBS VFGC feature. For more information see "Virtual Facilities Group Controls (VFGC)" on page 3-37.

### Message Detail Recording (MDR)

This feature is identical to the EBS MDR feature. For more information see "Message Detail Recording (MDR)" on page 3-40.

### Music on Hold

Music on Hold automatically provides music or announcements to calls placed on hold by the Camp On, Call Hold, Call Park, and Directed Call Park features. A caller placed on hold hears the music (or announcements) until either the caller hangs up or the DMS-10 switch takes the call off hold.

This feature is identical to the EBS MOH feature. For more information see "Music on Hold" on page 3-44.

### EBS Group Name and Number

This feature is identical to the EBS group name and number feature. For more information see "EBS Group Name and Number" on page 3-45.

### CLASS on Centrex

The EBS CLASS features are identical to the residential CLASS features. For more information see "Custom Local Area Signaling Services (CLASS)" on page 8-2.

### Call Forward Internet Down (CFID)

The EBS CFID feature is identical to the residential CFID feature. For more information see "Call Forward Internet Down (CFID)" on page 8-4.

### Call Reference Busy Limit (CRBL)

The EBS CRBL feature is identical to the residential CRBL feature. For more information see "Call Reference Busy Limit (CRBL)" on page 8-4.

# Packet Trunks

The DMS-10 supports a VoIP packet trunking feature. The VoIP trunks will use SIP signalling to interface virtual DMS-10 trunks to carriers using the internet. For more information see "Trunk features" on page 5-1.

# Miscellaneous Features

The following features can applied to residential or Centrex lines using VoIP terminals. The follows features may also apply to SIP packet trunks.

## Advanced Intelligent Network (AIN, CDP, FCD, OHD, OHI, TA)

For SIP lines, as dial tone is provided by the SIP client, the off-hook immediate (OHI) trigger will not be encountered until after the SIP user has completed dialing. The dialed digits are discarded.

For more information see "Advanced Intelligent Network" on page 10-1.

## CALEA

For more information see "CALEA Functional Overview" on page 11-1.

## Calling Line Identification (CLI)

For details about the Calling Line Identification, refer to overlay CLI in the NTP entitled *Data Modification Manual (297-3601-311)*.

## Emergency Service Bureau (ESB)

Because SIP user agents are portable and the exact location of the user is not readily available through SIP, two issues must be addressed:

- When a user temporarily moves location, a call to the emergency services will be connected to the emergency services bureau associated with the user's provisioned data (i.e., the emergency region assigned to the subscriber's directory number); this may not be the emergency services bureau closest to the caller's location.
- Caller identification data sent to the emergency services bureau is determined from the user agent's provisioned data (i.e., directory number). This information may reflect the user's permanent location and not the actual location of the caller when they have temporarily moved to another location.

For more information see "Emergency service bureau (ESB)" on page 6-24.

## Local Number Portability (LNP)

For more information see "Local Number Portability" on page 10-22.

## Multiple Appearance Directory Number

The Multiple Appearance Directory Numbers (MADN) feature allows the service provider to assign the same Directory Number to a VoIP line and a Time Division Multiplex (TDM) line.

Multiple Appearance Directory Number (MADN) Single Call Arrangement (SCA) allows a single DN to appear on multiple phones or MBS keys. Plain vanilla MADN SCA only allows a single member of the MADN group to be active at a time. If any one of the members is off hook then the other members will not be able to originate a call. If any of the MADN group members are off hook then anyone attempting to terminate on that DN will be sent to Busy Treatment. If the group is idle (no one is off hook or active on a call) and someone terminates on the MADN DN then all members of the group will ring. At this point any ONE of the MADN members can answer the call.

The Multiple Appearance Directory Numbers (MADN) feature allow a MADN DN to be used as a residential DN or as a member of a Centrex group (i.e., EBS or IBS station option). In Generic 602.10 and earlier, MADNs were only available to EBS group members. The number of MADN groups was also increased to 16,384.

## T.38 Fax

T.38 Fax signaling is supported for SIP lines and over SIP packet trunks. For SIP packet trunks, the DMS-10 will have the ability per trunk group to negotiate T.38 signaling for facsimile transmissions.

## Miscellaneous Line Features

### Alarm-checking Access (ALCK)

Allows the station access to the alarm checking feature.

### AMA Message (AMAM)

For any billable call originated by a subscriber that has been assigned the AMAM option, the AMA200 (DMS) / AMA201 (ATT) message will be output regardless of the print prompt assignment in overlay AMA (AMA) for that call type.

### Complaint-observed Study (COPL)

Provides a detailed record on AMA tape of message-rate customers' answered and unanswered recordable calls.

### Class-of-service Tone (COS)

Option is assigned to the originating station and provides a particular class-of-service tone to the terminating party of a call when the route over which the call is being placed is set up to determine the class-of-service mark based on station.

### Customer Assignable Options (OPTn)

For more information see "Customer Assignable Options (OPTN)" on page 4-4.

### Customer Assignable Station Options, enhanced (!x)

For more information see "Customer Assignable Station Options, enhanced (CASO)" on page 4-4.

### Denied Originating (DOR)

For more information see "Denied Originating (DOR)" on page 4-4.

### Denied Terminating (DTM)

For more information see "Denied Terminating (DTM)" on page 4-4.

### Emergency Region (EMR n)

The station is in emergency region n, where n is a number from 0 through 15.

### Foreign Exchange (FXA)

Only the FXA option can be assigned to a SIP station. For more information see "Foreign Exchange (FX) facility access" on page 6-38.

### Free Number Terminating (FNT)

For more information see "Free Number Terminating (FNT)" on page 4-5.

### Inter-LATA Restriction (IRST)

For more information see "Inter-LATA Restriction (IRST)" on page 6-21.

### Local Call Detail Recording (LCDR)

Provides a detailed record of all seven-digit calls originated by this station, that is, calling number, called number, duration of call answer, and disconnect entry.

### Long Duration Call Reporting Disabled (LDCD)

For more information see "Long Duration Call Reporting (LDCR)" on page 9-6.

### Override Thousands Group (OTHP)

A station that appears in a pre-subscribed thousands group may have the pre-subscribing option overridden by using the OTHP option.

### Personal Identification Number (PIN)

Specifies the digits that must be dialed to allow a caller access to feature manipulation through an access directory number.

### Presubscription options (PRES, PRS2, PRS3, PICL)

For more information see "Presubscription (PRES)" on page 6-20.

### Ring Again Denied (RAGD)

For more information see "Ring Again Denied (RAGD)" on page 3-18.

### Flat-rate, Message-rate and Message-rate Business (1FR, 1MR, 1MB)

SIP supports all three individual billing rates.

### Station Suspended (SUS, SUSO, SUST)

For more information see "Station Suspended Options (SUS, SUSO, SUST)" on page 4-6.

### Specific Carrier Restriction (CRST)

For more information see "Specific Carrier Restriction (CRST)" on page 6-21.

### Rate Treatment Package (RTP)

RTP is an originating characteristic of the station. RTPs are defined for each class of service and rate center, in Overlay AREA.

### Restricted Station Options (RES1, RES2)

For more information see "Restricted Station Options (RES1, RES2, LOCO)" on page 3-32.

> ***Note:*** *The LOCO option does not apply to Residential lines.*

### Service-observed Study (SOBS)

Allows for sample checks of end-to-end billing accuracy on answered recordable calls.

### Special Billing Number (SPB)

Toll calls for this station are billed to specific directory number.

### Subscriber Line Usage Study (SLUS, TSLS)

Used to determine measured service tariffs and to determine tariff effects for answered and unanswered calls.

### Toll Denied (TDN)

For more information see "Toll Denied (TDN)" on page 4-9.

### WATS services (IWT, OWTF, OWTF)

For more information see "INWATS (IWT)" on page 4-5. For more information see "OUTWATS (OWTF and OWTM)" on page 4-6.

### Traffic Separation Index Feature (STSI, DTSI)

Source and destination traffic-study options used for the Traffic Separation Measurement System (TSMS) feature.

# Section 9: Administrative features

## Introduction

This section briefly describes the administrative features offered by the DMS-10 switch.

### Security access

Each terminal and each password will be assigned one or more task-access rights. In order for any given task to be performed on a data terminal, the task-access rights of both the terminal and the user's TTY class password must match the access level required for that task. For additional information about the Security Access feature refer to the CNFG (PSWD) prompting sequence in the NTP entitled *Data Modification Manual* (297-3601-311).

Additional security can be provided by requiring a user account name and password to be entered in addition to the user's TTY class password when logging into a terminal. Refer to the CNFG (LOGU) prompting sequence in the NTP entitled Data Modification Manual (297-3601-311) for a description of the FLGI prompt.

### Remote data terminals

Terminals located at a site remote from the DMS-10 switch may be used to perform maintenance, data modification, and administrative functions from a central location.

### Interactive maintenance priority

Interactive maintenance priority minimizes contention for use of the overlay area by allowing higher priority programs to preempt lower priority programs. The classes of programs in order of priority are:

**1)** maintenance

**2)** traffic

**3)** data modification

### Multiple Overlay Access System

The Multiple Overlay Access System provides the capability to access more than one overlay program and to execute the programs concurrently.

The overlays are divided into two groups, administrative and maintenance. One overlay from the maintenance group and multiple, compatible administrative overlays can be active at the same time. Not only can different overlays be loaded at the same time, but the same overlay can be loaded multiple times from different I/O ports. This allows high usage overlays to be accessed at anytime without the inconvenience of having to wait for another user to finish their DMO activity. The administration overlays are described in NTP 297-3601-311, *Data Modification Manual*. The maintenance overlays are described in NTP 297-3601-506, *Maintenance Diagnostic Input Manual*.

## Data modification

Data base management in the DMS-10 switch is handled through a series of overlay programs that are used for initial office data generation and for ongoing data modification in response to subscriber service orders, and network requirements.

The overlays that cover data modification orders (DMOs) and file system update allow the operating company to:

- enter assignments of lines, directory numbers, trunks
- define routing, trunking, and translation of calls
- establish billing and charging information
- declare equipment configurations
- provide interfaces to Operations Support Systems (OSSs)

A description of the DMO overlays and data conversion can be found in the NTP entitled *Data Modification Manual* (297-3601-311).

### User-Friendly DMOS

If the user is uncertain about what are valid inputs to system prompts or what are valid options for diagnostic commands within an overlay, a “?” can be typed as a request for help. The user can request assistance in response to system prompts (commands and messages); the DMS-10 switch will provide a menu of acceptable inputs, then re-prompt for user selection. Incorrect responses are rejected, and the user is re-prompted until the response is correct or until the session is aborted.

## Operational measurements

The DMS-10 switch has programs, resident in system memory, that monitor system performance and level of service. These programs accumulate data associated with various system functions and print out that data as a set of operational measurements. This self-monitoring function of the DMS-10 switch is entirely controlled by software and requires no external equipment. For detailed information, see the NTP entitled *Operational Measurements* (297-3601-456).

### CPU real-time measurement

This feature provides the operating company with approximate measurements of central processing unit (CPU) real-time usage. These measurements are used to make estimates of remaining CPU capacity.

### OPM output block expansion

This feature enables the display of OPM block registers in either five-digit or six-digit format. The selection of digit format is made in Overlay OMC (OMC), in the response to prompt "DIG" (see NTP 297-3601-456, *Operational Measurements*).

### OPM enhancement

This enhancement enables telcos to prevent scoring calls initially on CAMA, CAM2, EAOS, EAS, EQA, ICP, ISUP, LEAS, LTRK, or PRI type routes that encounter busy condition and are re-routed to lockout state or to an alternate route. The telco controls this capability through the PGNC prompt in Overlay ROUT (ROUT).

## Digitone Fraud

Digitone (DGT) Fraud is primarily a cutover tool for offices converting to a DMS-10 switch. It provides dial tone to all stations and identifies those without the DGT option.

The Digitone Fraud feature is declared using data modification orders (DMOs). See the MTCE (Maintenance) prompting sequence of Overlay CNFG in the NTP entitled *Data Modification Manual* (297-3601-311). The table can be queried or cleared using Overlay OMC in the NTP entitled *Operational Measurements* (297-3601-456). A query request will provide a list of up to 64 station numbers. If more than 64 numbers are to be listed in the table, an overflow message appears, and any subsequent numbers are lost. Consequently, the user should check the table frequently and clear the table (using Overlay OMC) after printing the list of numbers.

## Line Load Control (LLC)

The Line Load Control (LLC) feature is used in extreme emergencies to restrict originating service to subscribers (such as police and fire departments, hospitals, and government agencies) who have been designated as "essential users." These users are assigned line circuits in positions 12 through 14 in each shelf of Peripheral Equipment, positions 26 through 31 in each line subgroup of Line Concentrating Equipment (even when located in the RLCM, OPM, OPAC, RSLE, RSLM, and RSC-S) or of a Remote Carrier Urban (RCU), the first four card slots (Positions 1-8; 25-32; 49-56; 73-80) of each shelf of a SLC for Subscriber Loop Equipment (SLE), or the first 128 lines of a RDT located on an ESMA. Other PE, LCE, SLE or ESMA positions are not service-protected and cannot use line circuits located in an ISDN line drawer. LLC does not apply to Virtual DNs. Virtual DNs will be able to originate calls when LLC is active.

Calls that are in progress and calls that are terminating are not affected when LLC is activated. Lines that are not service-protected remain without originating service and receive no indication other than no dial tone until LLC is deactivated.

LLC is activated or deactivated by using the administrative password and a maintenance terminal. In the event of a software system reload (SYSLOAD), LLC will be dropped and must be reactivated by way of a maintenance terminal. LLC will remain activated during an Initialization. For a list of LLC commands, refer to the resident command table in the NTP entitled *Maintenance Diagnostic Input Manual* (297-3601-506).

While LLC is activated, the message "LLC001 ACT" will be printed every 15 minutes.

## System data

Information on office and station data queries can be found in the NTP entitled *Data Modification Manual* (297-3601-311).

### Office data query

This feature allows a record of system data relating to directory numbers, lines, trunks, and memory to be requested and printed out at a maintenance terminal.

### Station data query

This feature allows operating-company personnel to obtain a maintenance terminal printout of office data associated with directory numbers and the status of both call-forwarding and speed-calling lines.

## Incremental office extension

Based on the operational and system data, operating-company personnel may add or remove peripheral equipment circuit packs in order to accommodate changes in line and trunk requirements. For detailed information, see the NTP entitled *Provisioning* (297-3601-450).

## Automatic Patch Reload (APR)

This feature applies to Generic 402.52 and later 400-Series generics. Automatic Patch Reload (APR) provides the capability of storing software patches on the file system and of automatically loading these patches into memory in the event of a SYSLOAD. APR is executed through commands in Overlay UPDT (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*).

## Opcode Patching

The Opcode Patching feature enables opcode patches to be applied automatically or manually to an XPM (RSC-S, SMS-R, or SCU) connected to a DMS-10 switch. “Automatic” patches are applied immediately after an XPM has been downloaded, without manual intervention. “Manual” patches are applied by entering the “APPL” command from Overlay DED; the command is entered for each XPM unit to which the patch is to be applied.

An opcode patch that has been applied can also be removed. Patches are removed by entering the “REMV” command from Overlay DED; the command must be entered for each XPM unit from which the patch is to be removed.

For a procedure used to apply or remove manual patches, see MP 1571 in NTP 297-3601-511, *Maintenance and Test Manual*.

## Line identification

For complete information on line identification, see the NTP entitled *Data Modification Manual* (297-3601-311).

### Calling Line Identification

This feature, which is activated through Overlay CLI, permits the operating company to determine the source of calls placed to any specific subscriber, whether or not served by the DMS-10 switch. When activated, the Calling Line Identification program collects and prints out data on all calls placed to the specified subscriber through the DMS-10 switch. CLI can be applied to a maximum of 32 stations.

#### Dedicated CLI Terminal

Dedicated CLI Terminal is a CLI feature enhancement that enables a TTY to be dedicated to receiving CLI print-outs. This new *CLI* class of message output may be assigned to the TTY in conjunction with other compatible message output classes. For a complete list of compatible message output classes, see NTP 297-3601-311, *Data Modification Manual*, Overlay CNFG (LOGU), prompt USER. For additional information concerning message output and message output classes, see Section 3 in NTP 297-3601-311, *Input/Output System*.

### Malicious Call Hold

Malicious Call Hold (or Calling Line Identification Hold) is used in conjunction with Calling Line Identification. When activated, it permits the specified subscriber (called party) to hold the connection on any call.

### Call Trace

The Call Trace feature can be activated through Overlay TRAC, which may be accessed after entering the administrative password at a maintenance terminal. Tracing will be allowed on the line(s) specified in this overlay, and can begin at either the originating or terminating end of a call. Call trace information can be output in terms of either trunk PE locations or directory numbers.

## Long Duration Call Reporting (LDCR)

The Long Duration Call Reporting feature enables operating companies to identify subscribers that regularly place long-duration calls, on an office-wide basis. This information enables the operating company to efficiently groom switch traffic by moving specific subscriber lines within the switch line equipment rather than by randomly selecting lines to move.

The feature operates by reporting the location of subscriber lines on which calls have exceeded a minimum call duration specified by the operating company. This information displays in either an LDC001 message, for originating calls, or an LDC002 message, for terminating calls. Display of the LDC001 and LDC002 messages is determined when the line register is idled. Thus, an LDC001 message can display for a terminating call if the identity of the line register changes from terminating to originating. Using Overlay CNFG (LDCR), the operating company defines the duration of a long-duration call and the number of LDC001 and LDC002 messages that can display within one hour.

When a long-duration call subscriber line has been identified, the operating company can assign to it a long duration call reporting disable (LDCD) option so that the LDC001/LDC002 messages can be suppressed for that subscriber line. Subscriber Line Usage Studies (SLUS) can then be used to obtain detailed information about usage for subscribers that have been assigned the LDCD option.

The following conditions pertain to the LDCR feature:

- MBS GIC keys are reported using the PDN of the MBS.
- MBS CPUG, DSS, and RAG keys report according to the key that actually processes the call.
- MBS CWT, 3WC, and UTF keys do not report long duration calls.
- Call reporting for MADN directory numbers is provided. The LDCD option needs to be assigned to each MADN location individually.
- Call reporting for TEEN directory numbers is provided. The primary directory number is printed in the LDC001/LDC002 message. The LDCD option is only assigned on the primary directory number to which TEEN has been assigned.
- Call reporting for multi-party line directory numbers is provided using only one of the directory numbers assigned to the line. The LDC001/LDC002 contains MP following the DN when a multi-party line is reported. The LDCD option may be assigned to any or all parties of a multi-party line. To eliminate the LDC001/LDC002 message, the LDCD option must be assigned to the DN in the display. The LDCD option can also be assigned to each of the other parties' DNs.
- Call reporting is not provided for PRI DNs.

- Call reporting is not provided for RCFA DNs.
- Call reporting is not provided for PCS subscribers.
- Call reporting is not provided for line trunk DNs.

## Custom Calling Tape Backup (CCTB)

Custom Calling Data (CCD) can be dumped to, or loaded from, the file system, using Overlay CCTB. When an Equipment Data Dump (EDD) is performed through Overly UPDT, overlay CCTB is loaded automatically. Overlay CCTB then dumps the CCD to the same file system device used for the EDD. If the Auxiliary Data Store (ADS) memory is subsequently lost during a SYSLOAD and initialization, overlay CCTB is again loaded automatically and CCD stored on the system I/O device is read into ADS memory. Overlay CCTB overlay can also be invoked manually. When configured, the Automatic Off-Site Database Backup (AODB) Feature will provide a location in the DMS-10 network that may be used to store copies of the Custom Calling Data. For information about the UPDT and CCTB overlays, see the NTP entitled *Maintenance Diagnostic Input Manual* (297-3601-506.).

## Automatic Off-Site Database Backup (AODB)

Automatic Off-Site Database Backup enables the operating companies to specify an additional location for the DMS-10 to access when performing office data dumps via overlays UPDT (Update) and CCTB (Custom Calling Tape Backup). This location will be defined by an Internet Protocol (IP) address in the DMS-10 network that will accept the office data files during the dump process. This IP location will be in addition to the IOI devices (HD0/HD1/MO0) already supported by the DMS-10. This allows operating companies the ability to create off-site archival copies of the office data, configuration record, and customer programmable custom calling data without having to travel to the individual DMS-10 sites to perform the data dumps.

The query function for both overlays UPDT and CCTB will be enhanced to show file creation information as part of the query output. The following information will be output during the query command: DMS-10 site name, year, month and day the file was created, hour and minute the file was created, and generic loaded when the file was created. In addition to the file creation information an indication as to which file is the active file in the sytem will also be output. The query output from overlay UPDT below shows there are two databases available on the queried IOI device. The configuration record and office data files associated with index 0 are the active files that would be used to restore the DMS-10 during a SYSLOAD. The STATUS field indicates this by showing those files as ACTV. The files associated with index 1 are from a previous data dump performed by UPDT as indicated by the STATUS field of INAC.

```
INDEX | SITE | MM/DD/YYYY HH:MM | GENERIC | STATUS
----------+------+--------------------------_----+-------------+------------
00        SYS1    06/22/2005       19:30    601.10       ACTV
01        SYS1    06/21/2005       19:36    601.10       INAC
```

The same output format is used in response to the query command for overlay CCTB. Data files stored at the IP location will also be queried and output with the new output format.

Additional commands are added to overlays UPDT and CCTB to send data to the IP location, retrieve data from the IP location, and specify which data files should be the active files in the DMS-10. Specifics on the new commands can be found in the NTP entitled *Maintenance Diagnostic Input Manual* (297-3601-506.).

## Defensive Programming (DP)

Defensive Programming enables operating company personnel to test new or modified translators before they are installed for call processing use. This is accomplished by restricting modification and testing to inactive, test copies of the active call processing translators.

When the DP feature is installed and operating company personnel need to create or modify a translator, the system creates an inactive test copy of the active original translator. Modifications can be made only to this <u>inactive</u> <u>tes</u>t copy of the translator. After completing the modifications, operating company personnel may test the inactive test copy using normal translations test procedures. After testing is complete, operating company personnel may then activate the test copy so that it can be used for call processing. When the test copy is activated, the original translator is made inactive and is stored in memory. Because the original version of a translator can never be modified, whether in the active or inactive state, operating company personnel are assured of a reliable backup in the event that the original version needs to be reactivated. The inactive original translator remains in memory until operating company personnel remove it. Upon the removal of the inactive original translator from memory, the newly-activated test version becomes the original version.

An Equipment Data Dump (EDD) stores all versions of a translator (test or original) in their latest status (active or inactive). A time stamp indicates when each translator version was last modified and activated.

## Automatic Time-of-Day Change

This feature automatically adjusts the clock for Daylight Savings Time. On the first Sunday in April, the clock is automatically set ahead 1 hour. On the last Sunday in October, the clock is automatically set back 1 hour. The time change occurs at 2:00 AM. If the office is configured for AMA, the *TMAD* function is performed; otherwise the *TIME* function is performed (see “Time and Date Commands” in “Resident Commands” in NTP 297-3601-506, *Maintenance Diagnostic Input Manual*). To set up the feature, see prompt DST in Overlay CNFG (MTCE) in NTP 297-3601-311, *Data Modification Manual*.

# Device Downloading

Downloadable packs are described in the following paragraphs.

## NTMX77 Flash Memory Download

The NTMX77 Flash Memory Download feature enables memory chips on the pack to be re-programmed by the host. This eliminates the potential for introducing an inappropriate boot load file into a DMS-10 switch in the event that the NTMX77 was downloaded with a DMS-100 load.

Two Flash Electrically Erasable Programmable Read Only Memory (EEPROM) chips are provisioned on the NTMX77 circuit pack. Each EEPROM contains 256 Kbytes of storage. When one of the EEPROMs is executing from RAM (called the *executable* EEPROM), the second (called the *loadable* EEPROM) stands by as a backup. In the event that the executable EEPROM becomes corrupted, the loadable EEPROM takes over operation and becomes the new executable EEPROM.

A download file is sent to the NTMX77 EEPROM after the “DNLD MX77" command is issued from Overlay DED (see NTP 297-3601-506, *Maintenance and Test Manual*). The download process comprises three basic steps: 1) the loadable EEPROM is erased; 2) the EEPROM is loaded with new software; 3) the content of the EEPROM is copied to the second EEPROM in the unit. If a software mismatch occurs during the process, an error message is output.

## XPM Fast Download

XPM downloading normally requires placing one-half of an SCM-10S or SCM-10U XPM module in man-made busy state and leaving the module in simplex mode for the duration of the loading process, thus creating the potential for losing operation should a fault occur on the in-service unit during the download. The XPM Fast Download feature enhances the downloading process for XPMs by utilizing memory storage in the NT7X05 Flash Memory pack provisioned in the XPM to significantly reduce the amount of time that the XPM must be in simplex mode.

The XPM Fast Download feature comprises two separate downloads: one download provides the NT7X05 pack with the load for the XPM; another download provides the instruction for the XPM to obtain the load from the NT7X05. To download the NT7X05 pack, operating company personnel enter the “DNLD 7X05" command in Overlay DED (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*). Since the NT7X05 maintains the download information despite loss of power or system reload, it must be downloaded only once for each generic upgrade. To then download an XPM from the NT7X05 pack, operating company personnel enter the “DNLD SCSC/SCUC” command in Overlay DED. If the XPM doesn't have a functional NT7X05 pack equipped when the download command is issued, the XPM is downloaded directly from the file system. If the NT7X05 pack doesn't contain the appropriate XPM load file when the download command is issued, the XPM is downloaded from the NT7X05 and a mis-match error message displays.

The following conditions pertain to the XPM Fast Download feature:

- If a SYSLOAD occurs during the download of either the XPM or the NT7X05, the download is aborted. After the SYSLOAD, the download must be re-initiated.
- XPM Fast Download is not supported for the RSC-S.
- Downloading time required for an NT7X05 can take up to 25% longer than an XPM. Thus, XPM Fast Download doesn't improve the overall downloading time required for site upgrades unless NT7X05s that already have the upgraded XPM program load burned into flash memory are installed at the time of the upgrade. Since, however, NT7X05s are downloaded when the associated XPM is in service, downloading can be performed at any time, with no system impact.
- Because the download information contained in the NT7X05 pack is program information only, it can be used for any XPM of the same family within the same generic.
- Once downloading has started, it may not be aborted until it has completed.
- A fault on the NT7X05 pack is not a factor when XPM activity is switched.

## DS-30A Download

The DS-30A Download feature enables memory chips on the DS-30A Interface pack (NT4T04AK, or later) to be downloaded with firmware code.

The memory on an NT4T04 pack is divided into two sections: a *resident* area and a *download* area. The *resident* area contains firmware permanently installed at the factory, while the *download* area can be downloaded at the customer site using the DS-30A Download feature.

To ensure that the NT4T04 contains the appropriate version of firmware when the pack is being returned to service, the software compares the system version table firmware numbers with the corresponding NT4T04 pack version numbers. If the table version numbers match either *resident* or *download* versions on the pack, the NT4T04 returns to service executing the matched version of firmware. If there is a version number mis-match, the pack remains in the out-of-service state. Operating company personnel may force the pack into service, regardless of the mis-match, by issuing the return-to-service command with the IMED (immediate) option.

To ensure that the appropriate version of firmware is downloaded to the NT4T04, the software compares the system version table firmware numbers with the corresponding version numbers in the file system. If the table version numbers match the *download* version in the file system, the download proceeds. If there is a version number mis-match, the pack is not downloaded. Operating company personnel may force the download, regardless of the mis-match, by issuing the download command with the IMED (immediate) option.

## MLI Download

The MLI Download feature enables memory chips on the MLI Interface pack (NT4T05AE, or later) to be downloaded with firmware code.

The memory on an NT4T05 pack is divided into two sections: a *resident* area and a *download* area. The *resident* area contains firmware permanently installed at the factory, while the *download* area can be downloaded at the customer site using the MLI Download feature.

To ensure that the NT4T05 contains the appropriate version of firmware when the pack is being returned to service, the software compares the system version table firmware numbers with the corresponding NT4T05 pack version numbers. If the table version numbers match either *resident* or *download* versions on the pack, the NT4T05 returns to service executing the matched version of firmware. If there is a version number mis-match, the pack remains in out-of-service state. Operating company personnel may force the pack into service, regardless of the mis-match, by issuing the return-to-service command with the IMED (immediate) option.

To ensure that the appropriate version of firmware is downloaded to the NT4T05, the software compares the system version table firmware numbers with the corresponding version numbers in the file system. If the table version numbers match the *download* version in the file system, the download proceeds. If there is a version number mis-match, the pack is not downloaded. Operating company personnel may force the download, regardless of the mis-match, by issuing the download command with the IMED (immediate) option.

## TDS Download

The TDS Download feature enables memory chips on the Tone and Digit Sender (TDS) pack (NT4T01CC, or later) to be downloaded with firmware code.

> ***Note:*** *Although NT4T01CA and NT4T01CB firmware versions may be displayed, the firmware on the packs is not downloadable.*

The memory on an NT4T01 pack is divided into two sections: a *resident* area and a *download* area. The *resident* area contains firmware permanently installed at the factory, while the *download* area can be downloaded at the customer site using the TDS Download feature.

To ensure that the NT4T01 contains the appropriate version of firmware when the pack is being returned to service, the software compares the system version table firmware numbers with the corresponding NT4T01 pack version numbers. If the table version numbers match either *resident* or *download* versions on the pack, the NT4T01 returns to service executing the matched version of firmware. If there is a version number mis-match, the NT4T01 is automatically downloaded from the file system so that it returns to service executing the matched version of firmware.

## IDC Download

The ISDN Drawer Controller (IDC), (NT6X54DA, or later) supports download and flash bank administrative features. The IDC enables flash memory chips to be downloaded with firmware code. The use of flash memory reduces costs associated with retrofitting surface mounted firmware on IDC packs and also increases system reliability since flash memory retains downloaded firmware when power to the circuit pack is lost. The IDC contains two flash memory banks. Both banks are used for program store. Normally both banks contain identical data. IDC download enables the firmware code to be downloaded from the file system into the IDC pack's inactive Flash Memory bank. The IDC must be in the man-made-busy (MMB) or the in-service (INS) state before executing the DNLD command. The system response to the DNLD command is a pass (IDC firmware and generic file system table numbers match) or fail (firmware and tape table numbers do not match) indication. The IMED option is necessary if intentionally downloading a firmware version that does not match the generic file system table numbers.

After downloading new IDC firmware, program code in the inactive bank becomes active when a SYSLOAD is performed, or after executing the switch memory bank (SWME) command. The SWME command causes the inactive bank to become the active bank, and the active bank to become the inactive bank. Because the command initializes IDC firmware, the IDC must be in the MMB or the system-made-busy (SMB) state before executing the SWME command. In most cases, the SWME command is used when downloading IDC firmware, independent of a generic upgrade. During a generic upgrade, the SYSLOAD command ensures that the active bank contains the new firmware version.

## IOI (NT3T90) Download

The IOI Download feature enables memory chips on the Input/Output Interface pack (NT3T90BE or later versions of the pack) to be downloaded with firmware code.

The memory on an IOI pack is divided into two sections: a *resident* area and a *download* area. The *resident* area contains firmware permanently installed at the factory, while the *download* area can be downloaded at the customer site using the IOI Download feature.

To ensure that the IOI pack contains the appropriate version of firmware when the pack is being returned to service, the software compares the system version table firmware numbers with the corresponding IOI pack version numbers. If the table version numbers match either *resident* or *download* versions on the pack, the IOI returns to service executing the matched version of firmware. If there is a version number mis-match, the pack executes the resident version. Operating company personnel may force the pack into service, regardless of the mis-match, by issuing the return-to-service command with the IMED (immediate) option.

To ensure that the appropriate version of firmware is downloaded to the IOI pack, the software compares the system version table firmware numbers with the corresponding version numbers in the file system. If the table version numbers match the *download* version in the file system, the download proceeds. If there is a version number mis-match, the pack is not downloaded. Operating company personnel may force the download, regardless of the mis-match, by issuing the download command with the IMED (immediate) option.

## Network Interface (NT8T04) Pack Download

The firmware on the NT8T04 pack resides in a single flash memory bank. Although the firmware load is initially resident on the pack, the firmware may also be loaded at an operating company's site.

A single firmware load provides both the DS-30A and MLI applications on the NT8T04 pack, but only one application can be supported by the pack at a time. The application to be supported by the pack is assigned by operating company personnel in Overlay NET (see NTP 297-3601-311, *Data Modification Manual*). Operating company personnel may load firmware on the pack through Overlay NED (see NTP 297-3601-506, *Maintenance Diagnostic Input Manual*).

In order to return an NT8T04 pack to service, the pack's flash memory bank must contain the required version of the firmware for the generic. Operating company personnel can, however, force an NT8T04 pack into service, running a version of firmware different from that required by the generic.

## PGIC Download

The PGIC Download feature enables flash memory chips on the Packet Gateway Interface Controller pack (NT6T01) to be downloaded with firmware code. All firmware running on a PGIC can be upgraded via a field download, including the board support package loader, the board support package, the operating system, the compressed file system image, and the application tar file. The flash memory on an NT6T01 pack has redundant sections for each component of the firmware download, and at any given time only one of the sections is allowed by software to be in a non-executable state (for example, partially erased, as happens during a download). This architecture makes the PGIC fully field-upgradeable, without introducing the possibility of rendering a PGIC unusable if a catastrophic system event occurs while a download is in progress.

To ensure that the NT6T01 contains the appropriate version of firmware when the pack is being returned to service, the software compares the NT6T01 firmware version number expected by the current generic and patch level with the actual firmware version number reported by the pack. If there is a version number mismatch, the pack remains in the out-of-service state. Operating company personnel may force the pack into service, regardless of the mismatch, by issuing the return-to-service command with the IMED (immediate) option.

## Firmware Download Replacement Field Tool

This feature provides telcos with a file system that contains three replacement firmware download storage areas. These storage areas can be used by the telco to store up to three different firmware loads to be used when a download is required.

To install a new download file in a file system configured with this capability, the telco first copies the download file into the file system. After the download file is activated, it can be used for downloading any affected equipment. A data dump is then performed to save indexing information and activation status for the new download file. Since the new download file does not replace the existing download file on the file system, the original download file can be restored on the downloaded unit, if required.

# Section 10: Advanced Intelligent Network and Local Number Portability

## Advanced Intelligent Network

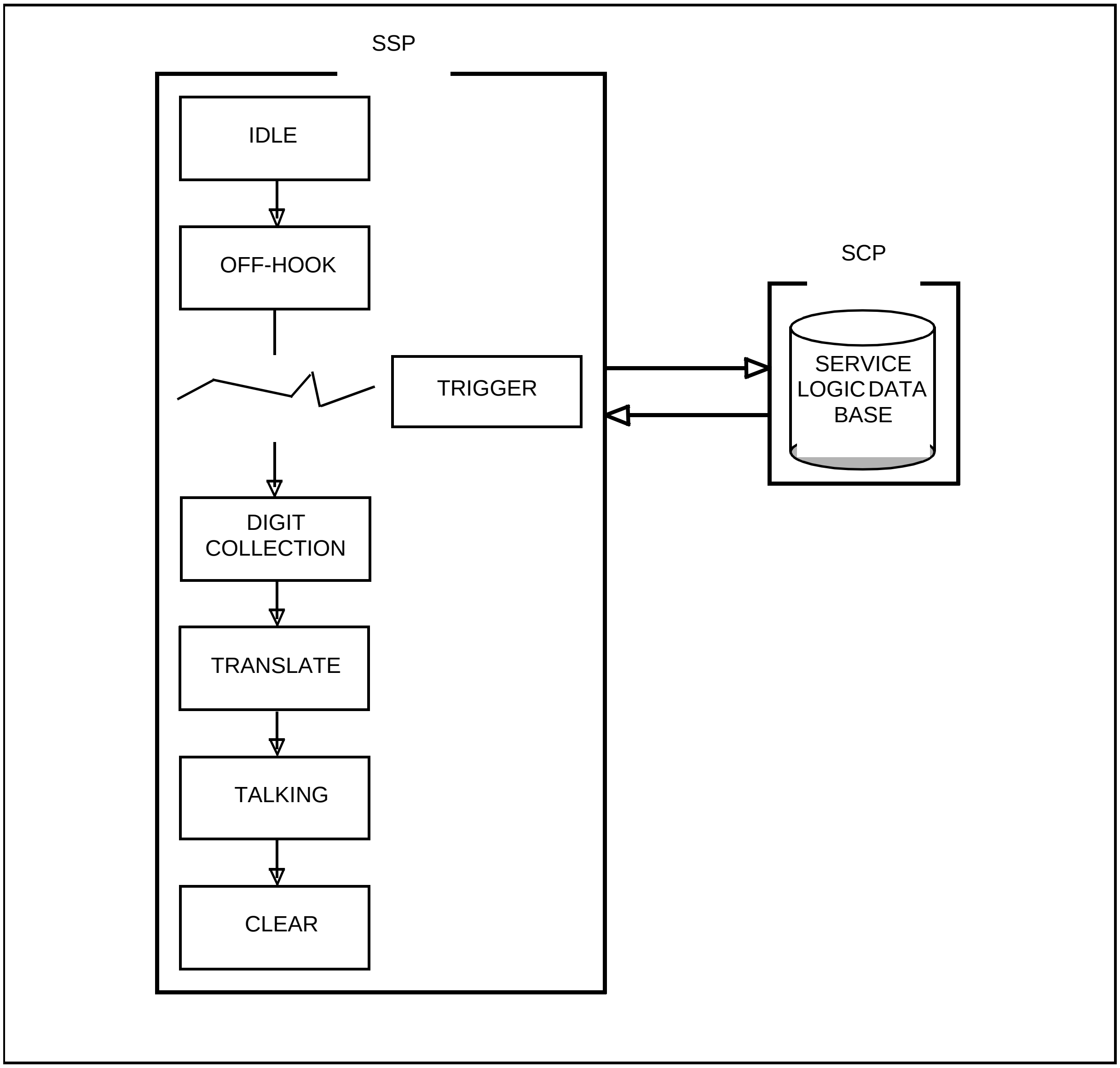
Advanced Intelligent Network (AIN) comprises several network elements that interact with switching systems to allow services to be defined outside of the internal call processing logic of the switching systems. With AIN feature development tools, the operating company can design and deploy features to its own specifications and make these features available across private and public networks. New subscriber features can be developed and implemented on the network without major disruption to current software loads of the individual network members.

AIN features enhance switch call processing logic capabilities to recognize calls that require additional processing and to query an application Service Control Point (SCP). Service logic programs at the SCP determine how AIN calls should proceed for further call processing. Queries and responses between offices equipped with AIN features and the SCP use Common Channel Signaling No.7 protocol.

### AIN functions

AIN processing is transparent to the subscriber. The majority of calls through the network access traditional functions that are standard features of the switch software. An AIN call signals the need for additional instructions through the use of specific events, or *triggers*. Triggers, such as an off-hook condition or specific dialed digits, indicate the need to interrupt call processing to request further processing instructions to complete the call. As shown in Figure 10-1, when a trigger is encountered by the switch software, a query is made to an application SCP for additional service logic to complete the call or implement a function. After the response to the query is returned, call processing resumes according to the instructions received, and the call is completed as any other DMS call.

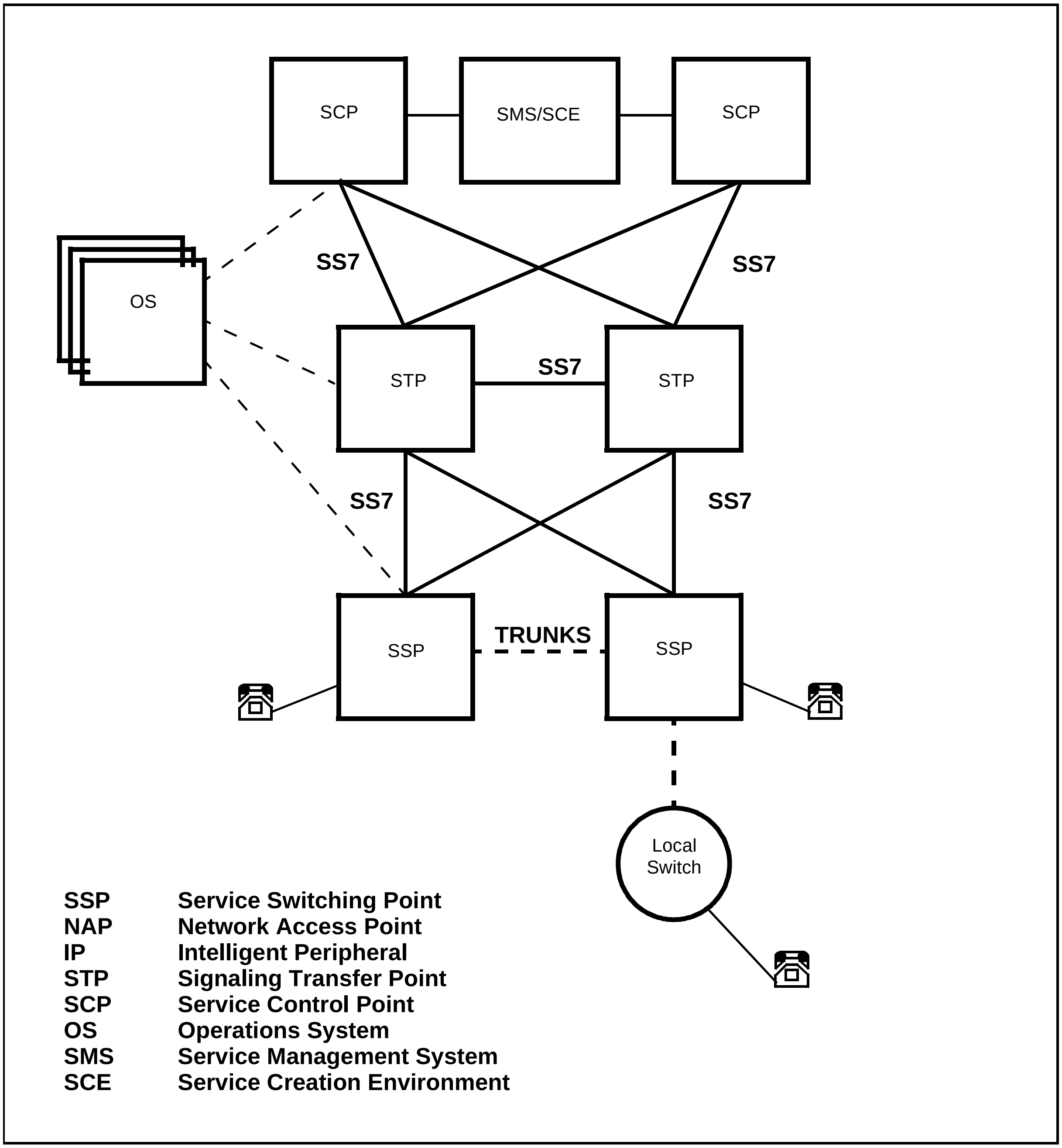
**Figure 10-1: Generic call processing model**



## AIN network architecture

AIN consists of a number of switching and processing nodes that are interconnected by signaling links. The basic AIN network components are shown in Figure 10-2.

**Figure 10-2: Advanced Intelligent Network architecture**



### Service Switching Point

Service Switching Points (SSP) are switching systems that are equipped with CCS7 hardware, software, and signaling links. SSPs provide subscriber access to the CCS7 network for routing and call control and CCS7-based services such as Custom Local Area Signaling Services (CLASS) and AIN.

For AIN, SSPs also identify calls associated with intelligent network services and initiate dialogues with the SCP in which the logic for the services resides. Upon detecting an AIN call, the SSP stops call processing and sends a TCAP query to the SCP. Upon receiving a response message, the SSP routes the call based on information received.

### Service Control Point

Service Control Points (SCP) contain the customer profiles, routing information, and service logic programs (SLP) needed to provide AIN service to subscribers. SSPs send signaling queries to SCPs, which in turn, send the necessary information for routing or feature execution back to the SSP. SCPs interface with SSPs through the CCS7 signaling network. SCPs may be deployed either on a regional basis, in a mated pair configuration, providing AIN service to multiple SSPs, or on a local basis.

### Signaling Transfer Point

Signaling Transfer Points (STP) are data switches that route CCS7 signaling information to the appropriate SSP, SCP, or STP. STPs interface only with the CCS7 signaling network.

### Local Switch

In the AIN network, a local switch is an end office that has no AIN capabilities. However, through software translations and class-of-service information, the local switch can route AIN calls to an SSP. Call control passes to the SSP.

### Operations System

An Operations System (OS) is a class of functions that provide operations, administration, maintenance, and provisioning (OAM&P) capabilities for network elements, network systems, software, and services.

### Service Management System

A Service Management System (SMS) is one of several OSs that may be used in the AIN architecture. The SMS is a centralized network administration system that performs management functions specifically for the SCP. The SMS enables operating companies to provision and administer AIN services and customer profiles required by the SCP. The SMS interacts with the SCP and has no direct communication with the SSP.

### Service Creation Environment

Creating and customizing AIN services is performed in the Service Creation Environment (SCE). The SCE interacts with the SCP and has no direct communication with an SSP. The SMS provides the SCE. New services are constructed, configured, and managed through the SMS.

## AIN Signaling

The queries and responses that pass between SSPs and SCPs are transported through CCS7 network equipment using CCS7 protocol. The elements of the CCS7 network that are used for processing AIN calls are described below. For a more detailed description of the CCS7 network, see Section 8 in this NTP.

### CCS7 protocol architecture

CCS7 operations are divided into a number of layers. The layers applicable to AIN operation include: Message Transfer Part (MTP), Signaling Connection Control Part (SCCP), and Transaction Capabilities Application Part (TCAP).

**Message Transfer Part (MTP)** The MTP manages signaling message routing by providing reliable transport and delivery of signaling messages across the CCS7 network.

**Signaling Connection Control Part (SCCP)** The SCCP provides additional functions to the MTP such as enhancing CCS7 routing capability by providing application addressing and management. It also keeps track of the status of applications and informs the user when an application is unavailable. In the DMS-10 switch, the SCCP is used to request Global Title Translations (GTT) and then to transport AIN messages.

**Transaction Capabilities Application Part (TCAP)** The TCAP is used to control non-circuit related information transfer between two or more signaling nodes through the CCS7 signaling network. The TCAP supervises message transactions between an SSP and an SCP.

The SCCP provides information transfer functions between applications and the MTP. The SCCP transfers TCAP signaling messages to the signaling network without setting up a signaling connection. TCAP uses the SCCP in order to transport data to another node, for example, a message to an STP requesting GTT.

**Service logic host route** A service logic host route (SLHR) is used for addressing, either directly or indirectly (for example, through GTT), an SCP. Information defined within the SLHR is also used to format a TCAP query message. Up to fifteen SLHRs may be assigned per DMS-10 office.

For indirect addressing, the SLHR includes the following information:

- destination point codes (DPC) of the mated STP pair that will handle the GTT
- translation type to be used by the STP to route the message to the appropriate SCP application
- global title source code, which indicates what data to use in populating the global title value in the TCAP message: charge number, called party identifier, or default source (charge number or called party ID, depending on the trigger)

For direct addressing, the SLHR includes the following information:

- DPC of the SCP

## AIN basic call model

A basic call model for AIN shows the points during call processing at which the SSP may suspend the call in response to a trigger and query an SCP for instructions on how the call is to be handled. The AIN basic call model consists of two parts: the originating portion of a call and the terminating portion of the call. Both call model parts are partitioned into blocks known as *points-in-call*. A point-in-call describes the call processing performed and the information that has been gathered at that particular stage in the call. Figure 10-3 shows the basic call model for AIN call origination. Figure 10-4 shows the basic call model for AIN call termination. Tables 10-A and 10-B define the points-in-call in the AIN call models.

**Figure 10-3: AIN Originating Basic Call Model**

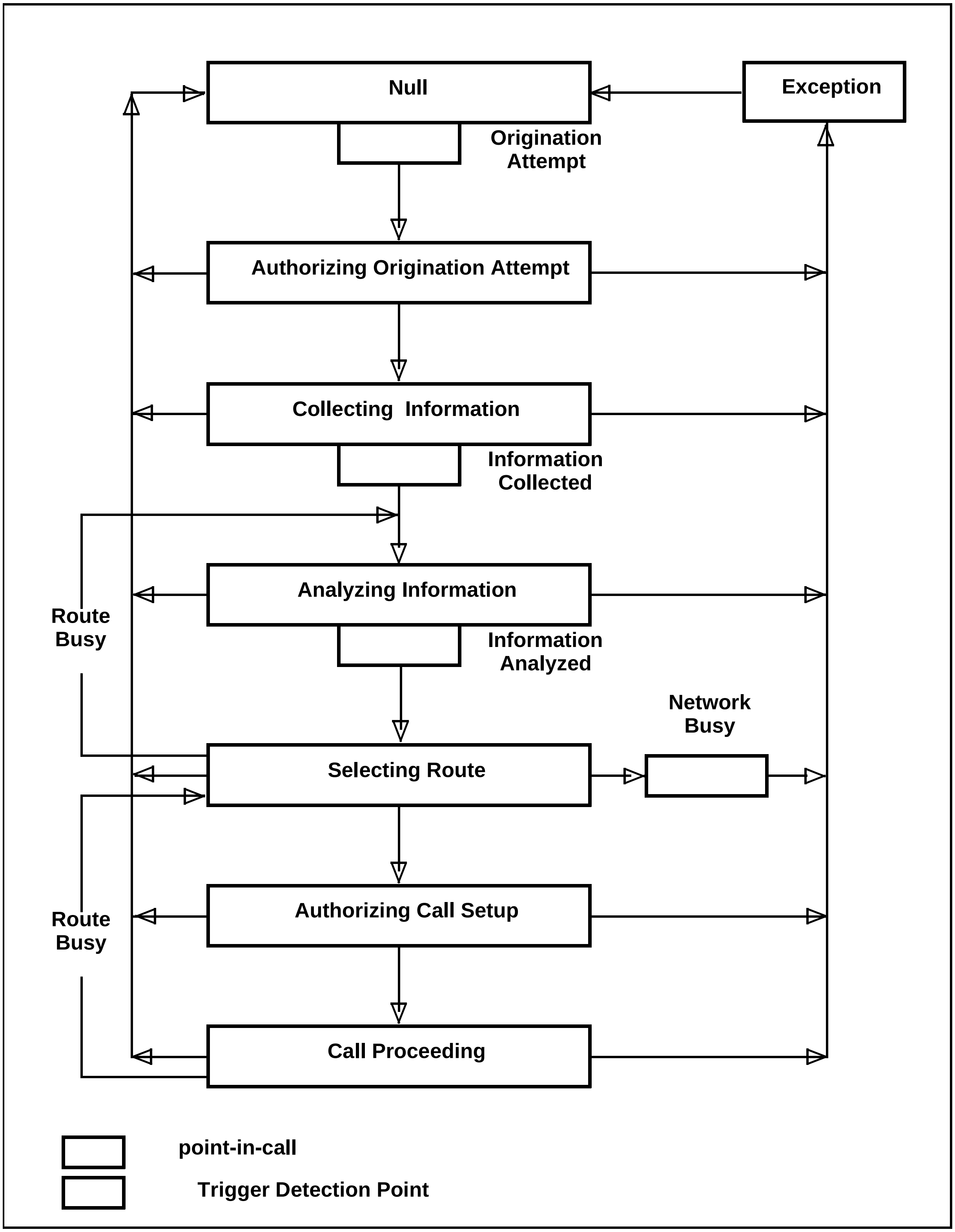
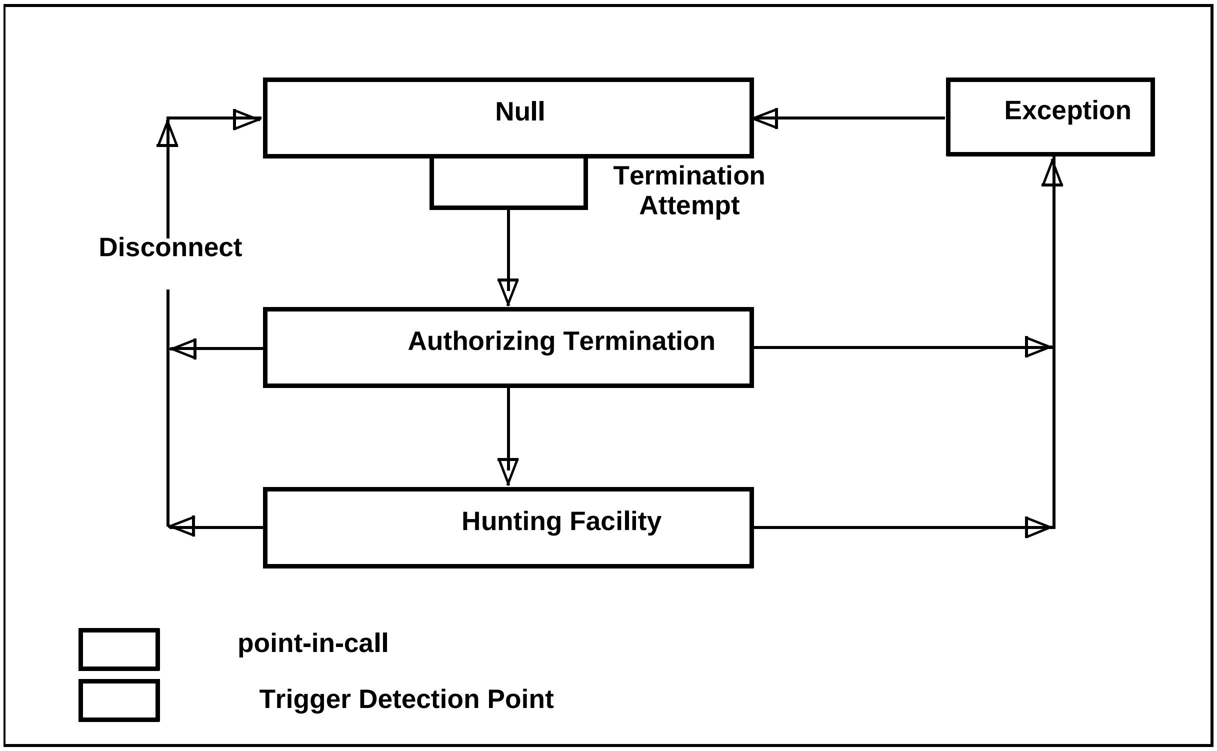


**Figure 10-4: AIN Terminating Basic Call Model**




| **Table 10-A: AIN Origination Basic Call Model points-in-call** | |
|---|---|
| **Point-in-call** | **Description** |
| Null | The call is idle and no connection exists. The agent goes off-hook and attempts to originate a call. |
| Authorizing Origination Attempt | The authority of the originating agent to place a call is verified. |
| Collecting Information | Complete initial address information is collected from the originating agent (for example, access code, dialed digits). |
| Analyzing Information | Digits collected are translated according to the agent's dial plan. Called party ID, type of call, and carrier are determined. |
| Selecting Route | The SSP attempts to route the call to the route specified in routing tables or to the directory number. |
| Authorizing Call Setup | The SSP verifies the authority of the calling party to place the call (for example, toll denied). |
| Call Proceeding | The SSP sends a message to the terminating resource indicating a desire to set up the call. |

| Table 10-B: AIN Termination Basic Call Model points-in-call | |
|---|---|
| **Point-in-call** | **Description** |
| Null | The call is idle and no connection exists. A message from the originating resource indicates a call to be set up. |
| Authorizing Termination | The SSP verifies the authority to route the call to the terminating resource. |
| Hunting Facility | The busy/idle status of the terminating agent is determined. |

## Trigger detection points

In the call models, *trigger detection points* (TDP) specify those points in a call at which the SSP determines if an SCP is to be queried. A trigger is activated when the following conditions are met:

- the call has reached a trigger detection point
- the trigger is assigned
- trigger criteria are met
- the trigger is active

When a trigger is activated, call processing is interrupted and a query is sent to an SCP to obtain additional information in order to complete the call or implement a function. The service logic program (SLP) resident at the SCP determines how the AIN call is to be handled and sends a response to the SSP. After the response is received at the SSP, call processing resumes according to the instructions received, and the call is completed.

Several triggers may be associated with one trigger detection point. A single call may cause several triggers to activate; the maximum number of trigger activations for a given call is specified by the operating company.

Trigger detection points, triggers, and the criteria for trigger activation are described below and are summarized in Table 10-C.

### Origination attempt TDP

This TDP occurs in the AIN originating call model at the null point-in-call. Only the off-hook immediate trigger is applicable to this TDP.

**Off-hook immediate trigger** This trigger is designed to formulate and send a query to an SCP as soon as an associated line goes off-hook.

An example showing how the trigger might be used to prevent unauthorized usage of telephones during off-hours is illustrated in the following call scenario. In the call scenario, subscriber A is assigned the off-hook immediate trigger. The SCP maps this trigger for this particular station to the Authorized Usage feature. The SCP deactivates the off-hook immediate trigger at the start of normal business hours and activates the trigger at the end of normal business hours. Callers are automatically disconnected when attempting to make a call during off hours.

The call proceeds as follows:

**1)** Subscriber A goes off-hook.

**2)** The call proceeds to the origination attempt trigger detection point. The SSP verifies that subscriber A is assigned the off-hook immediate trigger and that the trigger is activated. The SSP sends a query package that includes the calling number to the SCP.

**3)** The SCP examines the calling number, creates a response and sends the response that instructs the SSP to disconnect subscriber A.

**4)** The SSP disconnects subscriber A by routing the call to the AIND (AIN disconnect) generic condition.

The off-hook immediate trigger is available on ISDN BRI terminal service profiles (TSP).

### Information collected TDP

This TDP occurs in the AIN originating call model at the collecting information point-in-call. This TDP is encountered after complete initial address information becomes available. There are two triggers applicable to this TDP: off-hook delay trigger and shared interoffice trunk trigger.

**Off-hook delay trigger** This trigger applies to all calls (including international calls) except when an *escape code* has been dialed or the dialed digits correspond to the activation of, deactivation of, or access to a switch-based feature. An *escape code* provides the subscriber a means of making certain calls (for example, 911) without activating the off-hook delay trigger. An escape code list is maintained at the SSP. These codes must be either 0-, 00-, a valid North American Numbering Plan number, or a PODP N11 service code.

An example showing the use of the off-hook delay trigger for toll restriction is illustrated in the following call scenario. In the call scenario, subscriber A is assigned the off-hook delay trigger. The SCP maps this trigger for this particular station to the Toll Restriction feature. When a toll call is made, whether a 1+, 0+, or 10XXX DN is dialed, the SCP examines a list of NPAs and thousands groups that are toll-free from this station. For calls that are blocked (that is, the called number is not on this list), the calling party has the ability to enter a Personal Identification Number (PIN) to override the call blocking.

The call proceeds as follows:

1) Subscriber A dials a 1 + 10-digit pattern.
2) The SSP collects the digits, and when the call proceeds to the information collected trigger detection point, verifies that Subscriber A is assigned the off-hook delay trigger and sends a query to the SCP that contains the called number and the calling number.
3) The SCP compares the called number with the toll-call list and returns a response to the SSP that instructs the SSP to play an announcement and collect a four-digit PIN.
4) The SSP plays the appropriate announcement and, after subscriber A dials a four-digit PIN, sends a message to the SCP that includes the dialed PIN.
5) The SCP verifies the PIN and returns a message that includes the original called number to the SSP.
6) The SSP translates the digits sent by the SCP and routes the call accordingly.

The off-hook delay trigger is available on ISDN BRI terminal service profiles (TSP).

**Shared interoffice trunk trigger** This trigger may be assigned to an incoming or two-way public trunk group which supports Equal Access Multi-Frequency or ISUP signaling. This trigger allows the SSP to provide AIN capabilities to an EAEO.

The trigger is activated for all calls originating on the assigned trunk group that match the defined 0ZZ-XXXX. The DMS-10 switch uses the XXXX information to determine that SSP processing is required and 0ZZ to identify the call as an AIN call.

An example showing the use of the shared interoffice trunk trigger to provide an end office with AIN capability, even if the end office does not have CCS7 connectivity or AIN software, is illustrated in the following call scenario. In the call scenario, the end office wants to allow carrier selection to be decided at the SCP, based on time-of-day, for certain subscribers. The end office sets up translations to route a call to the access tandem using a specific trunk group, specifying the 0ZZ-XXXX, when a subscriber with CASO (Customer Assignable Station Option) dials a 1+ call.

The call proceeds as follows:

1) Subscriber A dials a 1+ number.
2) The EAEO sets the specific 0ZZ-XXXX information and out-pulses using the dialed digits.
3) The SSP collects the signaling information and, when the call encounters an activated shared interoffice trunk trigger, sends a query to the SCP that includes the trunk group ID and the dialed digits.

**4)** Using the time-of-day and trunk group ID, the SCP selects a carrier and sends a response to the SSP that instructs the SSP in routing the call.

**5)** The SSP uses the information sent by the SCP to route the call using the specified carrier.

### Information analyzed TDP

This TDP occurs in the AIN originating call model at the analyzing information point-in-call. There are four triggers applicable to this TDP: public office dialing plan (PODP) 3 through 10-digit trigger, public office dialing plan (PODP) feature code trigger, Customized dialing plan trigger, and public office dialing plan (PODP) N11 trigger.

**Public office dialing plan (PODP) 3 through 10-digit (DIG) trigger** This trigger enables lines and trunks to directly trigger on a number in the North America Numbering Plan (NANP). If the user dials seven digits, then the NPA digits are added and triggering is attempted based on 10 digits. If the user dials a carrier access code plus 10 digits (for example, 101XXXX+1+10 digits), the carrier access code is stripped off and triggering is based on the remaining 10 digits. In all cases, an attempt is made to convert the number dialed into a 10-digit national number and the to use this number as the trigger.

Triggering may occur on NPA, NPA-N, NPA-NX, NPA-NXX, through NPA-NXX-XXXX, provided that the number is in the NANP format.

An example of the use of the PODP 3 through 10-digit trigger is illustrated in the following call scenario. In the call scenario, a major rental company wants to advertise one directory number, 736-8227 (RENTCAR). This directory number together with the NPA can be used as a 10-digit trigger.

The call proceeds as follows:

**1)** Subscriber A dials 736-8227.

**2)** The SSP collects the digits, and after determining that the PODP DIG trigger is activated due to the digits dialed plus the implied 919 NPA, sends a query to the SCP that contains the calling number and the called number.

**3)** The SCP uses the calling number to determine the location of the nearest car rental office and sends a response message to the SSP containing the DN to be used for call completion.

**4)** The SSP translates the digits returned in the SCP response message and routes the call accordingly.

**Public office dialing plan (PODP) feature code trigger** This trigger enables residential and MVP customers to generate an AIN transaction when a specified feature access code (*XX(X) or 11XX(X)) digits are dialed. This trigger is assigned by DN or DN/call type. Customers with the appropriate parameter assigned to their lines are assigned all of the AIN services offered through feature access codes in their dialing plan.

An example showing the use of the PODP feature code trigger in conjunction with the Speed Call feature is illustrated in the following call scenario. In the call scenario, subscriber A is assigned the PODP feature code trigger. The SCP maps this trigger for this particular station to the Speed Call feature.

The call proceeds as follows:

1) Subscriber A dials *23.
2) The SSP collects the digits and, after the PODP feature code trigger is activated, sends a query to the SCP that contains the feature code and calling number.
3) The SCP uses the calling number and the feature code to identify the speed call number and sends a response back to the SSP that contains the DN necessary for call completion.
4) The SSP translates the digits returned in the SCP response message and routes the call accordingly.

The PODP feature code trigger is available on ISDN BRI terminal service profiles (TSP).

**Customized dialing plan trigger** This trigger enables IBS customers to generate an AIN transaction when the appropriate feature access code (*XX(X) or 11XX(X)), public network escape code (direct outward dialing access code such as 9+), or intercom (that is, station-to-station or group intercom) digits are dialed. This trigger is assigned only to IBS groups.

An example of the use of this trigger in conjunction with the Scheduled Forwarding feature is illustrated in the following call scenario. In this call scenario, Subscribers A and B are in the same customer group. This group is assigned the Customized Dialing Plan trigger. The SCP maps this trigger for this particular station to the Scheduled Forwarding feature.

The call proceeds as follows:

1) Subscriber A dials 8227.
2) In the SSP's translations, intercom code 8227 is assigned as a customized dialing plan trigger. When the trigger is activated, the SSP sends a query to the SCP that contains business group, calling number, and intercom digits.

3) The SCP uses the business group and intercom code, together with the time-of-day, to determine subscriber B's location (DN). The SCP sends a response message back to the SSP that contains the DN to be used for call completion.

4) The SSP translates the digits returned in the SCP response and routes the call accordingly.

**Public office dialing plan (PODP) N11 trigger** PODP N11 indicates that the call as dialed is an PODP N11 call, where N is a number from 2 through 9. This trigger is activated when lines and trunks dial the appropriate PODP N11 number. The PODP N11 trigger is office-based. The trigger is not applicable to international calls.

An example of the use of the PODP N11 trigger in conjunction with a Directory Assistance call is illustrated in the following call scenario. Directory Assistance (411) calls are routed to an Automated Directory Assistance Call Completion (ADACC) center as determined by the calling party's area code.

The call proceeds as follows:

1) Subscriber A dials 411.

2) The SSP collects the digits and, after the PODP N11 trigger is activated, sends a query to the SCP that contains the calling number and the PODP N11 digits (411).

3) The SCP uses the calling number to determine the trunk group to which the call should be routed and sends the information back to the SSP in a response message.

4) The SSP routes the call to the ADACC center in accordance with the information sent by the SCP.

### Termination attempt TDP

This TDP occurs in the AIN Terminating call model at the null point-in-call. Only the termination attempt trigger is applicable to this TDP.

**Termination attempt trigger** With this trigger, a query is sent to an SCP when a call attempts to terminate to the DN, DN/call type, or RCFA. The trigger is assigned to non-ISDN lines (on a DN basis), to ISDN BRI lines (on a DN/call type basis), and to RCFAs.

An example of the use of this trigger in conjunction with the Scheduled Call Forwarding feature is illustrated in the following call scenario. In this scenario, a business has locations on the east and west coast. The east coast office lines are assigned the termination attempt trigger. The SCP maps this trigger for these lines to the Scheduled Call Forwarding feature. Calls arriving at the east coast office after 5:00 PM Eastern time must be forwarded to the west coast office to take advantage of the different time zones.

The call proceeds as follows:

**1)** Subscriber A dials the DN of the east coast office.

**2)** The SSP translates the call, terminates to the station dialed by subscriber A and, after the termination attempt trigger is activated, sends a query to the SCP that contains the called number and calling number.

**3)** The SCP uses the called number and time-of-day to determine the DN to which the call is to be forwarded and sends this information in a response message to the SSP.

**4)** The SSP translates the digits returned by the SCP and routes the call accordingly.

### Trigger assignment (subscription)

Triggers may be assigned to non-ISDN lines (on a per-DN basis), to ISDN Basic Rate Interface (BRI) lines (on a DN/call type basis), to line-trunks (on a per-DN basis), to Remote Call Forward Appearances (RCFA), to public trunk groups, and to IBS groups. Triggers may also be assigned on an office basis. Supported trigger assignment is summarized in Table 10-C.

**Table 10-C:**
**AIN Trigger Activation Criteria, Associated Trigger Detection Points, and Assignment**

| Trigger Detection Point | Trigger | Criteria (dialed digits) | Subscription |
|---|---|---|---|
| Origination attempt | off-hook immediate | None | DN,<br>DN/call type |
| Information collected | off-hook delay<br><br><br><br><br>shared interoffice trunk | Not escape code in PODP.<br><br>Not access to switch-based feature.<br>Trunk signaling information | DN,<br>DN/call type<br><br><br><br>Public trunk group |

### Trigger precedence

When a call is at the information analyzed trigger detection point, several triggers may become activated. The SSP then verifies trigger assignment in the following order:

**1)** line or trunk group

**2)** customer group

**3)** office

If there are multiple PODP DIG triggers or customized dialing plan triggers that match the digits the subscriber has dialed, then the most significant pattern takes precedence. For example, a 10-digit number takes precedence over a 6-digit number, which in turn takes precedence over a 3-digit number.

An ISDN subscriber on a BRI line may be assigned any trigger that can be assigned to the DN/call type.

## AIN call processing

Typical AIN call progression includes the following steps:

**1)** After a call is placed by a subscriber, the SSP identifies any triggers that have become activated by the call, indicating that the call requires AIN processing. If a trigger has been activated, the SSP suspends normal call processing.

**2)** The SSP checks for code gapping controls. Code gapping controls are set when the SCP or Adjunct detects a near overload condition and sends a message to the SSP indicating that queries are to be temporarily suspended. If code gapping controls apply, the SSP gives the call final treatment.

**3)** If code gapping controls do not apply, the SSP creates a Transaction Capability Application Part (TCAP) query message and sends the message to the SCP over the CCS7 network.

**4)** The SCP creates a response message and sends the message to the SSP. The SSP performs the instructions sent in the message which may include routing the call, providing special terminating treatment, or continuing call processing using the information already collected. The instructions may also include playing an announcement or collecting more dialed digits.

### AIN recorded announcements

The operating company provides equipment used for AIN service announcements. Communication between the DMS-10 SSP and the recorded announcement units is supported using existing DMS-10 trunk signaling (MF only). When an announcement is requested, the DMS-10 SSP seizes an idle trunk from the trunk group linked to the announcement unit by sending an off-hook signal to the announcement unit. After the announcement unit returns a wink signal to the SSP, the communication link between the two systems is established. The SSP then sends streams of MF digits containing announcement-related information to the announcement unit. The SSP cuts through a connection between the announcement unit and the calling subscriber. At the end of the announcement, the announcement unit goes on-hook. If there is no interruption from the SSP, the announcement unit goes off-hook after one second and repeats the announcement. This process repeats until the SSP clears the connection or requests a different announcement.

When a non-interruptible announcement is played, the SSP clears the connection with the announcement unit either when the calling subscriber hangs up, after the announcement is played, or after a message requesting cancellation is received from the SCP. When an interruptible (“play and collect”) announcement is played, the SSP clears the connection either after the calling subscriber hangs up, after the announcement is played, or after the calling subscriber dials a digit.

### Call example

The following call example, illustrated in Figure 10-5, shows AIN call processing activity in detail using the public office dialing plan (PODP) N11 trigger example described earlier. In this example, the subscriber is assigned the off-hook delay trigger, the digit sequence, 411, has been defined by the local office as an escape code, and the PODP N11 trigger (411) has been set up for the office.

The call proceeds as follows:

**1)** The subscriber goes off-hook. The call proceeds to the origination attempt trigger detection point (TDP). The SSP determines that the subscriber is not assigned the off-hook immediate trigger and the call proceeds to the authorizing origination attempt point-in-call.

**2)** At the authorizing origination attempt point-in-call, the authority for the subscriber to make the call is verified.

**3)** The call proceeds to the collecting information point-in-call and the 411 digits are then collected.

**4)** The call proceeds to the information collected TDP. The SSP checks to determine whether the subscriber is assigned the off-hook delay trigger. Because the subscriber is assigned this trigger, the SSP checks to determine whether an escape code or switch-based feature code was dialed. Since 411 has been defined by the office as an escape code, the trigger is not activated and the call proceeds to the analyzing information point-in-call.

**5)** At the analyzing information point-in-call, the dialed digits have already been translated according to the subscriber's dialing plan. Additional digits may be required if the dialed number matches any of the PODP 3 through 10-digit triggers set up for the office. In this example, no more digits are required.

**6)** The call proceeds to the information analyzed TDP. Because the PODP N11 trigger is assigned on an office-wide basis and because 411 was dialed, the trigger is activated.

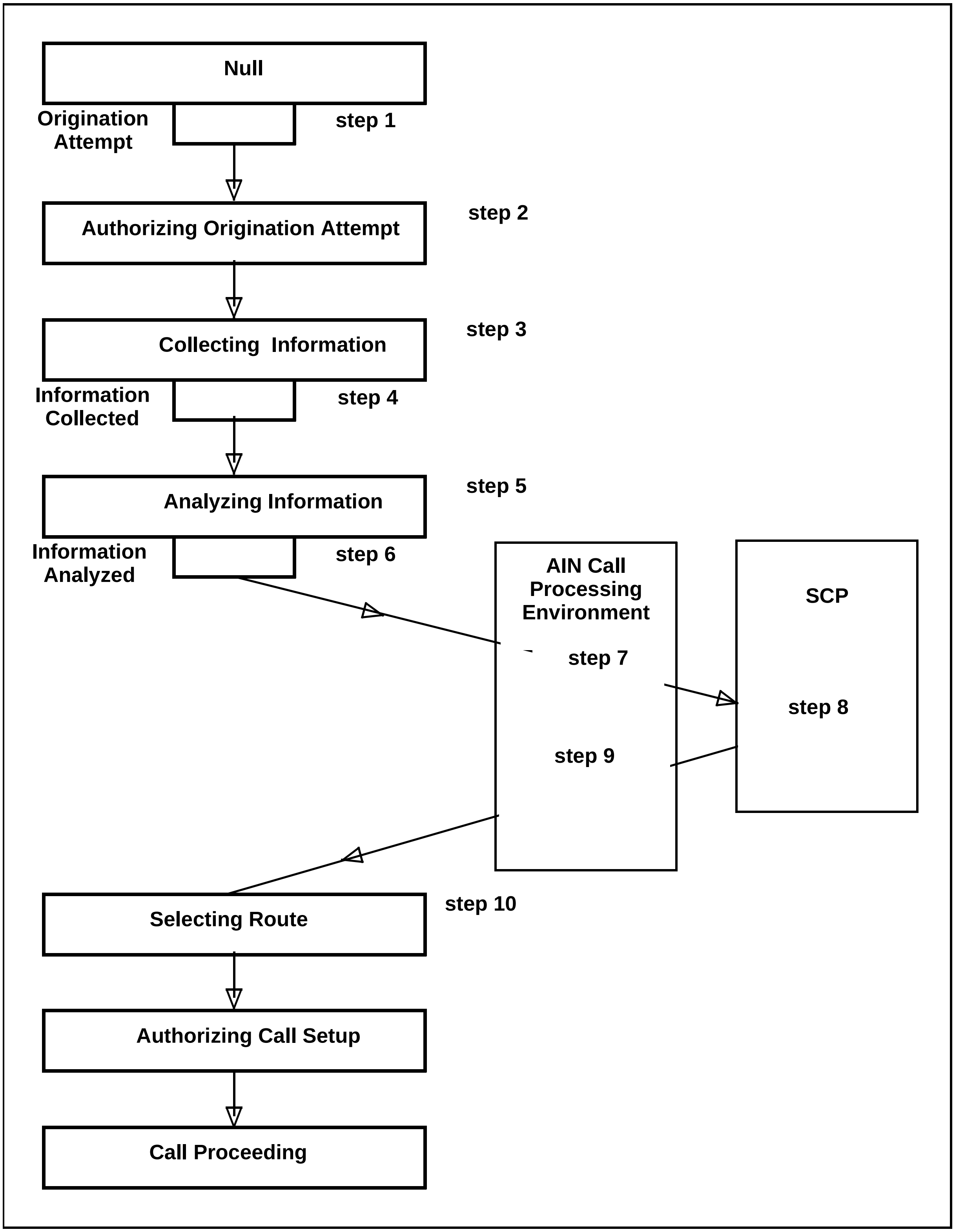
**7)** In response to the information analyzed TDP activation, the AIN call processing environment identifies service logic host route information and verifies that network blocking controls are not in effect. The SSP sends a TCAP query message to the SCP and suspends processing for the call.

**8)** The SCP receives the query message and then determines that the “411" service logic program should be used for the call. The program instructs the SCP to return a message to the SSP containing routing information.

**9)** The SSP receives the message from the SCP and decodes the message. The AIN call processing environment examines the trunk group routing index and updates the routing information with the new routing information received from the SCP.

**10)** The SSP resumes call processing at the selecting route point-in-call. Call processing uses the new routing information to route the call and connect the subscriber to the directory assistance bureau.

**Figure 10-5: AIN Originating Call Example - PODP N11 Trigger**



### AIN Termination Notification

The AIN Termination Notification feature enables the SSP to report Termination Notification to the SCP when it receives a Send Notification request. The Termination Notification message contains termination information about an AIN call after the call has reached a NULL point-in-call (PIC) state due to either the call being answered (followed by disconnect), the called line being busy, abandonment of the call, an exception condition, or an unrelated error condition. This information can be used to determine whether the call completed at the SSP, to analyze timing of call duration, to bill AIN calls at the SCP, and to conduct performance for query-response activity at the SCP.

**Processing overview** The feature processes Send Notification requests in the following manner:

**1)** The SSP receives a Send Notification component following a Call Related component (Analyze Route, Authorize Termination, Continue, Forward Call, or Send to Resource) in the same response package from the SCP.

**2)** When the Send Notification request is received, the AIN call processing application performs the following activity:

-look for answer and termination events

-detect answer of the call and start call timing

-detect termination of the call and send back to the SCP a Termination Notification message containing either the termination reason if the call is not answered or the duration of the call if the call is answered.

Multiple Send Notification requests can be processed in a single call, up to a maximum of one for each of the six triggers that a call may encounter. A single Termination Notification message is sent in response to each individual request as illustrated in the following call scenario example:

**1)** An AIN call encounters a PODP 10-digit trigger and the SSP launches a query to the SCP.

**2)** The SCP sends back a Call Related Continue response component with a Send Notification request.

**3)** The call continues and encounters a PODP 6-digit trigger. The SSP launches a second query to the SCP.

**4)** The SCP sends back a an Analyze Route response component with a Send Notification request.

**5)** The call eventually exits to the NULL PIC state and separate Termination Notification messages are sent to the SCP in response to each Send Notification request.

A separate Termination Notification message is sent to the SCP in response to a Send Notification request, for each leg of a forwarded call or three-way call.

## Network Management

The SSP provides overload controls to prevent the SCP from being flooded with messages in the following two ways:

- SCP overload control
- Service Management System Originated Code Control (SOCC)

The SCP invokes automatic code gapping controls by sending the SSP a message that indicates the type of control (SCP overload control or SOCC) to apply, the gap interval, the control duration time, and the control code. After receiving and processing the message contents, the SSP routes new AIN calls matching the criteria specified by the SCP to reorder tone or announcement. After the automatic code gapping timer expires, AIN calls are processed in the normal manner. The SSP is able to simultaneously control up to 64 codes on the AIN SCP overload control list and 64 codes on the AIN SOCC overload control list.

Both SOCC and SCP overload controls can be directed at originating stations or dialed digits (outgoing destination, including local terminating) depending on the nature of the service and type of trigger.

## AIN operations, administration, maintenance, and provisioning

AIN is a software feature that uses existing DMS-10 family hardware and CCS7 communication links. The operating company supplies the SCP and recorded announcement equipment. The CCS7 software package is a prerequisite for using TCAP messaging between the DMS-10 SSP and the SCP.

### Billing

The DMS-10 Automatic Message Accounting (AMA) recording system automatically collects and records billing information for calls served by the DMS-10 switch. For a complete description of AMA processing pertaining to AIN, see NTP 297-3601-124, *Automatic Message Accounting System*.

### Operational measurements

The DMS-10 switch provides operational measurements on which maintenance, traffic, accounting, and provisioning decisions are based. This information is provided on demand both to local and to remote maintenance centers, and in regularly-scheduled reports.

Operational measurements for AIN are collected for each trigger, for calls from subtending offices that have encountered AIN triggers, and for network-related maintenance. For a complete description of the operational measurements pertaining to AIN, see NTP 297-3601-456, *Operational Measurements*.

### Maintenance and diagnostics

Standard DMS-10 switch maintenance and operation procedures are used to test the operation of any systems or circuits used by the AIN feature. Maintenance programs may be scheduled to run automatically or on request. Only regular maintenance activity associated with the DMS-10 switch is required. For information about AIN maintenance requirements, see NTP 297-3601-500, *General Maintenance Information*.

### System engineering

For information about AIN engineering requirements, see NTP 297-3601-450, *Provisioning*.

# Local Number Portability

The Local Number Portability (LNP) feature enables subscribers to retain their directory number (used only for circuit-switched calls) when changing either to a different telephone service provider or to a different physical location, within the same rate center. For example, a subscriber may switch from a traditional wire-line carrier to a new local service provider such as a cable TV operator; even though the subscriber's DN does not change, all calls placed to the subscriber must then be routed to the new service provider's switch. Or, subscribers may elect to continue to be serviced by a service provider even though the subscriber has moved into an area normally served by another switching complex.

## Number porting intra-LATA calls

When subscribers elect to change to a different service provider (*recipient exchange*), their NPA-NXX-XXXX is identified in their current service provider's (*donor exchange*) data base as being vacant or ported out. All switches in the rate center are then updated to reflect that the NPA-NXX is now “portable.”

To uniquely identify switches in the same rate center, each switch is assigned a 10-digit number called a *location routing number* (LRN). At the SCP, ported DNs are associated with the LRNs of switches to which subscribers are being ported. Thus, when an NPA-NXX that has been ported is dialed, the SCP is queried and returns a response containing the LRN, which is used (through translations) to route the call to the recipient exchange.

Switches that do not have SCP access capability route the call either to the donor switch or to a tandem switch that has SCP access capability. The donor or tandem switch, then, queries the SCP to determine routing for the call.

## Number porting inter-LATA calls

Inter-LATA calls involving ported numbers are carried by inter-LATA exchange carriers (IXC). Because an IXC must be able to recognize the NPA-NXXs of LATAs where it maintains a point-of-presence (POP), the IXC performs SCP queries and then routes calls to the appropriate recipient exchanges.

## LNP trigger

The need for additional instructions for an LNP call is signaled through the use of an LNP trigger. Although the LNP trigger is assigned like the PODP 3 through 10-digit trigger, it is a conditional trigger and supports only from six through ten digits. A query is sent to the SCP only if additional call processing conditions are satisfied after the trigger is initially encountered.

LNP triggers can be assigned along with PODP 3 through 10-digit triggers. If an LNP trigger and a PODP 3 through 10-digit trigger have matching or overlapping digit patterns, the PODP trigger has precedence.

### LNP trigger conditions

An SCP query is not sent under the following conditions:

- the called party has an associated line card on the local switch or the party is reachable indirectly (for example, through a thousands group pointed to a trunk group associated with a PBX)
- a query has already been performed for this call, or the serial triggering limit has been reached
- the call is an inter-LATA call (call routing through an IXC is required)

LNP triggering is bypassed (no SCP query is performed) if 0-, 0+, service codes (411, 911, etc.), or service access codes (800, 900, etc.) are dialed.

### Query/response processing

If all LNP trigger criteria have been satisfied, the SCP is queried. The query message is an AIN info_analyzed message with parameters populated in the same manner as for the PODP 3 through 10-digit trigger.

The response from the SCP may consist of the following:

- analyze_route: the associated called party (called party ID) contains the LRN associated with the ported number or the called party number (non-ported number) populating the info_analyzed query message. This is the number that is retranslated.
- continue
- disconnect

If the SCP is unavailable, or if the SCP response contains a fatal error indicator, then the call is routed as if the trigger had not been encountered.

### Translation and routing

The digits returned in an analyze_route response from the SCP are re-translated, which results in one of the following:

- the call terminating within the switch
- the call being routed directly to the recipient exchange
- the call being routed to the serving exchange by way of an intermediate exchange

To route LNP calls between offices, four ISUP parameters are used in the IAM: called party number parameter (CdPN); generic address parameter (GAP); forward call indicator (FCI) M-bit (translated called number indicator); and the jurisdiction information parameter (JIP). If an LRN is returned in the SCP response to an SCP query, the LRN is stored in the CdPN, the GAP contains the dialed digits, and the FCI M-bit indicates that an LNP query has taken place. The JIP contains the first six digits of the LRN and identifies the office of the calling party for billing purposes. The JIP is an optional ISUP parameter that is sent only if indicated in the outgoing trunk group's data.

## LNP line trigger

The LNP line trigger allows an SCP query to be performed when the DN dialed is assigned as a station or RCFA (Remote Call Forward Appearance) on the same switch as the call originator. The LNP line trigger allows operating company personnel to pre-provision station data for subscribers that are porting into the switch. It also allows subscriber station data to remain on the switch that a subscriber is porting out of. Thus, the move of a subscriber from one switch to another is synchronized since the SCP knows the correct location of the DN.

When the SCP is downloaded with the new location of the DN, the LNP line trigger may be removed from the station data for the ported-in DN, and the station data on the switch that the subscriber ported out of may be deleted. These changes do not, however, have to be performed immediately after the SCP has been downloaded with the new location of the DN. The DN may exist on two different switches at the same time.

The LNP line trigger is assigned to a DN through station option LNPT in Overlay DN (STN), for POTS lines, or DN (DNCT), for ISDN lines. The trigger can also be assigned to the following intra-switch groups of lines that are normally ported together:

- EBS groups (through prompt LNPT in Overlay HUNT (EBS))

  ***Note:*** *Beginning in Generic 412.20, the LNPT station option can be assigned to a DN in an EBS group through Overlays DN (STN), for POTS lines, or DN (DNCT), for ISDN lines.*

- IBS groups (through station option LNPT, in Overlay DN (STN), assigned to each station in the group)
- DNH groups (through prompt LNPT in Overlay HUNT (DNH))

  ***Note:*** *Beginning in Generic 412.20, the LNPT station option can be assigned to a DN in a DNH group through Overlays DN (STN), for POTS lines, or DN (DNCT), for ISDN lines.*

- TEEN lines (through station option LNPT, in Overlay DN (STN), assigned to the PDN and inherited by the SDN)

### LNP line trigger SCP query/response processing

Table 10-D shows LNP Line Trigger SCP query and response processing.

**Table 10-D:**
**LNP Line Trigger SCP query/response processing**

| Cases | Former Service Provider | | Former Service Provider | |
|---|---|---|---|---|
| | **Prior to SCP update** | **After SCP update** | **Prior to SCP update** | **After SCP update** |
| Porting for the first time | Dialed DN is returned - call termination on the local switch | LRN is returned - call is routed to new service provider | Dialed DN is returned - call is routed to the donor | Home LRN is returned - call terminates on the local switch |
| Previously ported - porting again | Home LRN is returned - call terminates on the local switch | New LRN is returned - call is routed to the new service provider | Former LRN is returned - call is routed to the former service provider | Home LRN is returned - call terminates on the local switch |
| Previously ported - porting back to the donor | Home LRN is returned - call terminates on the local switch | Dialed DN is returned - call is routed to the donor | Former LRN is returned - call is routed to the former service provider | Dialed DN is returned - call terminates on the local switch |

## Signaling

The DMS-10 switch can act as an originating, intermediate, or recipient exchange. The following tables show possible signaling scenarios for the DMS-10 switch.

### DMS-10 switch as an originating exchange

Table 10-E shows the possible signaling scenarios for a DMS-10 switch acting as an originating exchange.

In scenario number 1, the LRN is received in the SCP response. The LRN is translated, which results in the call being routed to an ISUP trunk. The IAM message parameters are populated as follows: the CdPN contains the LRN, the GAP contains the dialed digits, and the FCI M-bit is set to “number translated.”

In scenario number 2, the dialed DN is received in the SCP response. The dialed DN is retranslated in the translations for LNP, which results in the call being routed to an ISUP trunk. The IAM message parameters are populated as follows: the CdPN contains the dialed number, there is no GAP, and the FCI M-bit is set to “number translated.”

In scenario number 3, no LNP query is performed. Normal translations routes the call over an ISUP trunk. The FCI M-bit is set to “number not translated.”

In scenario number 4, the LRN is received in the SCP response. The LRN is retranslated in the translations for LNP, which results in the call being routed to an inband trunk. The dialed number is outpulsed over the inband trunk.

**Table 10-E:**
**Originating exchange signaling**

| # | LNP trigger response | Basis for routing | Routing tables | Outgoing signaling | | | |
|---|---|---|---|---|---|---|---|
| | | | | Type | FCI M-bit | GAP | CdPN |
| 1 | LRN | LRN | LNP | ISUP | Translated | DN | LRN |
| 2 | DN | DN | LNP | ISUP | Translated | no GAP | DN |
| 3 | No trigger | DN | Normal | ISUP | Not translated | no GAP | DN |
| 4 | LRN | LRN | LNP | MF | N/A | N/A | DN |
| 5 | DN | DN | LNP | MF | N/A | N/A | DN |
| 6 | No trigger | DN | Normal | MF | N/A | N/A | DN |
| 7 | SCP failure | DN | LNP | ISUP | Not translated | no GAP | DN |
| 8 | SCP failure | DN | LNP | MF | N/A | N/A | DN |

In scenario number 5, the LNP response contains the dialed DN. The DN is retranslated in the translations for LNP, which results in the call being routed over an inband facility, and thus over an inband trunk.

In scenario number 6, no LNP query is performed and the call is routed through normal translations to an inband trunk.

In scenario number 7, the SCP cannot be accessed. By default, the call is translated in the translations for LNP, which results in the call being routed to the switch on which the NPA-NXX is located. The IAM message parameters are populated as follows: the FCI M-bit is set to “number not translated,” there is no GAP, and the CdPN contains the dialed DN.

In scenario number 8, the SCP cannot be accessed. By default, the call is translated in the translations for LNP, which results in the call being routed to the switch on which the NPA-NXX is located. The dialed number is outpulsed over the inband trunk.

### DMS-10 switch as an intermediate exchange

Table 10-F shows the possible signaling scenarios for a DMS-10 switch acting as an intermediate exchange. An intermediate switch can perform SCP queries and is able to process and route incoming inband and ISUP traffic. Calls are routed using both normal translations and translations for LNP.

In scenario number 1, the intermediate switch receives an LNP response with another switch's LRN. The LRN is retranslated in the translations for LNP to determine an appropriate outgoing route. If the outgoing route is ISUP, the CdPN parameter is populated with the LRN, the GAP is populated with the DN, and the FCI M-bit is set to “number translated.”

In scenario number 2, the intermediate switch receives an SCP response containing a dialed DN. The dialed DN is retranslated in the translations for LNP in order to determine an appropriate outgoing route. If the outgoing route is ISUP, the CdPN parameter is populated with the dialed number, the GAP is not included, and the FCI M-bit is set to “number translated.”

In scenario number 3, the incoming trunk is inband and the switch determines that an SCP query is not needed. The FCI M-bit defaults to “number not translated” and the GAP is not included in the outgoing IAM.

In scenario number 4, the intermediate switch receives an SCP response containing another switch's LRN. The LRN is retranslated in the translations for LNP in order to determine an appropriate outgoing route. If the outgoing route is to an inband facility, the dialed number is outpulsed to the next exchange.

In scenario number 5, the intermediate switch receives an SCP response containing a dialed DN. The dialed DN is retranslated in the translations for LNP in order to determine an appropriate outgoing route. If the outgoing route is an inband facility, the dialed number is outpulsed to the next exchange.

**Table 10-F: Intermediate exchange signaling**

| # | Incoming signaling | | | | LNP trigger response |
|---|---|---|---|---|---|
| | **Type** | **FCI M-bit** | **GAP** | **CdPN** | |
| 1 | MF | N/A | N/A | N/A | LRN |
| 2 | MF | N/A | N/A | N/A | DN |
| 3 | MF | N/A | N/A | N/A | No trigger |
| 4 | MF | N/A | N/A | N/A | LRN |
| 5 | MF | N/A | N/A | N/A | DN |

**Table 10-F: (Continued)**
**Intermediate exchange signaling**

| # | Incoming signaling | | | | LNP trigger response |
|---|---|---|---|---|---|
| | **Type** | **FCI M-bit** | **GAP** | **CdPN** | |
| 6 | MF | N/A | N/A | N/A | No trigger |
| 7 | ISUP | Not translated | don't care | don't care | LRN |
| 8 | ISUP | Not translated | don't care | don't care | DN |
| 9 | ISUP | Not translated | don't care | don't care | No trigger |
| 10 | ISUP | Not translated | don't care | don't care | LRN |
| 11 | ISUP | Not translated | don't care | don't care | DN |
| 12 | ISUP | Not translated | don't care | don't care | No trigger |
| 13 | ISUP | Translated | DN | not LRN | N/A |
| 14 | ISUP | Translated | N/A | not LRN | N/A |
| 15 | ISUP | Translated | N/A | not LRN | N/A |
| 16 | ISUP | Translated | DN | not LRN | N/A |
| 17 | ISUP/MF | Not translated | N/A | N/A | SCP failure |
| 18 | ISUP/MF | Not translated | N/A | N/A | SCP failure |

| Basis for routing | Routing tables | Outgoing signaling | | | |
|---|---|---|---|---|---|
| | | **Type** | **FCI M-bit** | **GAP** | **CdPN** |
| LRN | LNP | ISUP | Translated | DN | LRN |
| DN | LNP | ISUP | Translated | no GAP | DN |
| DN | Normal | MF | Not translated | no GAP | DN |
| LRN | LNP | MF | N/A | N/A | DN |
| DN | LNP | MF | N/A | N/A | DN |
| DN | Normal | ISUP | N/A | N/A | DN |
| LRN | LNP | ISUP | Translated | DN | LRN |
| DN | LNP | ISUP | Translated | no GAP | DN |
| DN | Normal | MF | Not translated | Pass | DN |
| LRN | LNP | MF | N/A | N/A | DN |
| DN | LNP | MF | N/A | N/A | DN |
| DN | Normal | MF | N/A | N/A | DN |
| LRN | LNP | MF | N/A | N/A | DN from GAP |
| DN | LNP | MF | N/A | N/A | DN |
| DN | LNP | ISUP | Translated | N/A | DN |
| LRN | LNP | ISUP | Translated | DN | LRN |
| DN | LNP | ISUP | Not translated | no GAP | DN |

| Basis for routing | Routing tables | Outgoing signaling | | | |
|---|---|---|---|---|---|
| | | **Type** | **FCI M-bit** | **GAP** | **CdPN** |
| DN | LNP | MF | N/A | N/A | DN |

In scenario number 6, the intermediate switch receives the dialed DN from an inband facility. The dialed DN is not in a portable NPA-NXX, therefore no LNP query is performed and the call is routed through normal translations using the dialed DN. Normal translations routes the call to an outgoing inband facility. The dialed DN is outpulsed over the inband trunk. LNP has no impact in this scenario.

Call processing in scenario number 7 is the same as that for scenario number 1.

Call processing in scenario number 8 is the same as that for scenario number 2.

In call scenario number 9, the intermediate switch receives an IAM through ISUP, but no SCP query is required. The call proceeds through normal translations and, because neither the dialed DN nor LRN is served by the switch, the call is routed out of the office over an ISUP trunk. The outgoing GAP and/or FCI M-bit are passed as they were received from the incoming IAM. If the called party is changed at the intermediate switch due to feature interaction, the FCI M-bit is set to “number not translated” and the GAP is removed from the IAM.

Call processing in scenario number 10 is the same as that for scenario number 4.

Call processing in scenario number 11 is the same as that for scenario number 5.

In scenario number 12, the intermediate switch receives an IAM through an ISUP trunk. The CdPN in the IAM is not a portable NPA-NXX and thus no SCP query is required. The call is processed through normal translations using the CdPN and is routed to an outgoing inband facility. The CdPN is outpulsed over the inband trunk as it is received from the incoming trunk. LNP has no impact on this scenario.

In scenario number 13, incoming signaling is ISUP. The IAM received contains the GAP and the FCI M-bit set to “number translated”; the CdPN does not contain the LRN of the intermediate switch. Since the outgoing signaling is inband, the switch sends the “ported number” GAP to the next exchange.

In scenario number 14, the intermediate switch receives an IAM with the FCI M-bit set to “number translated” and no GAP. Therefore, the switch does not perform an SCP query. The dialed number is in a ported NPA-NXX, but the number has not been ported. Thus, the intermediate switch routes the call through the translations for LNP using the CdPN.

In scenario number 15, the intermediate switch receives an IAM through ISUP, but an SCP query is not required. The dialed DN is not served by the switch and is routed out of the office over an ISUP trunk using the translations for LNP. The outgoing GAP and/or FCI M-bit are passed as they are received in the incoming IAM. If the called party is modified at the intermediate switch due to a feature activation, the FCI M-bit is set to “number not translated” and the GAP is removed.

In scenario number 16, the incoming signaling is ISUP. The received IAM contains the GAP, the CdPN does not contain the LRN of the intermediate switch, and the FCI M-bit is set to “number translated.” Since the LRN is not served by the switch, the call is routed out of the office using normal translations, over an ISUP trunk.

In scenario number 17, the intermediate switch attempts to perform an SCP query, but either there is an SCP failure or invalid data is returned in the SCP response. The call is routed through the translations for LNP.

In scenario number 18, the intermediate switch attempts to perform an SCP query, but either there is an SCP failure or invalid data is returned in the SCP response. The call is outpulsed over the MF trunk.

### DMS-10 switch as a recipient exchange

Table 10-G shows the possible signaling scenarios for a DMS-10 switch acting as a recipient exchange. The terminating switch has a list of unique North American numbering plan numbers that are defined as LRNs for this switch. When a call is incoming to this switch over an ISUP trunk and the FCI M-bit in the IAM is set to “number translated,” the switch first determines whether the called number is an LRN it is assigned. If the number is an LRN assigned to the switch, the “ported number” GAP replaces the LRN as the called number. This new called number is then used for routing the call.

**Table 10-G: Recipient exchange signaling**

| # | Incoming signaling | | | | LNP trigger response | Basis for routing | Routing tables |
|---|---|---|---|---|---|---|---|
| | **Type** | **FCI M-bit** | **GAP** | **CdPN** | | | |
| 1 | MF | N/A | N/A | Dialed DN | Dialed DN | Dialed DN | LNP |
| 2 | MF/ Line | N/A | N/A | DN | No trigger | Dialed DN | Normal |
| 3 | ISUP | Translated | none | DN | N/A | Dialed DN | Normal |
| 4 | ISUP | Translated | Dialed DN | DN | N/A | DN from GAP | Normal |
| 5 | ISUP | Not translated | N/A | DN | Switch's LRN | Dialed DN | LNP |

For incoming MF trunks, the CdPN is not an LRN since originating and/or intermediate switches that are capable of accessing the SCP do not send LRNs over an MF trunk. In the first scenario, the CdPN is not found on the switch so an SCP query is performed. If the query response includes the CdPN, the call is routed using the CdPN and the translations for LNP.

In the second scenario, the CdPN is found on the switch. An SCP query is not performed. The call proceeds through normal translations.

In scenario number 3, incoming signaling is ISUP. The FCI M-bit is set to “number translated.” Since a GAP was not received in the IAM, the CdPN does not contain an LRN. The CdPN is used as the call proceeds through normal translations. In this scenario, the originating switch has queried the SCP and has determined that the DN associated with the CdPN has not been ported.

In the fourth scenario, incoming signaling is ISUP. The FCI M-bit is set to “number translated,” a GAP is included in the IAM, and the CdPN contains the LRN of the terminating switch. The DN obtained from the GAP is used in normal translations to route the call.

In scenario number 5, the incoming signaling is ISUP. The FCI M-bit is set to “number not translated” and, thus, the CdPN does not contain the LRN. If the CdPN is not found on the switch, an SCP query is performed. If the subsequent response contains an LRN owned by the switch, the LRN is dropped and the call is routed using the CdPN and LNP translations.

## LNP call processing

The following call processing examples show how the local exchange carrier handles LNP calls.

### Call placed to a ported DN through a direct ISUP trunk to the recipient exchange

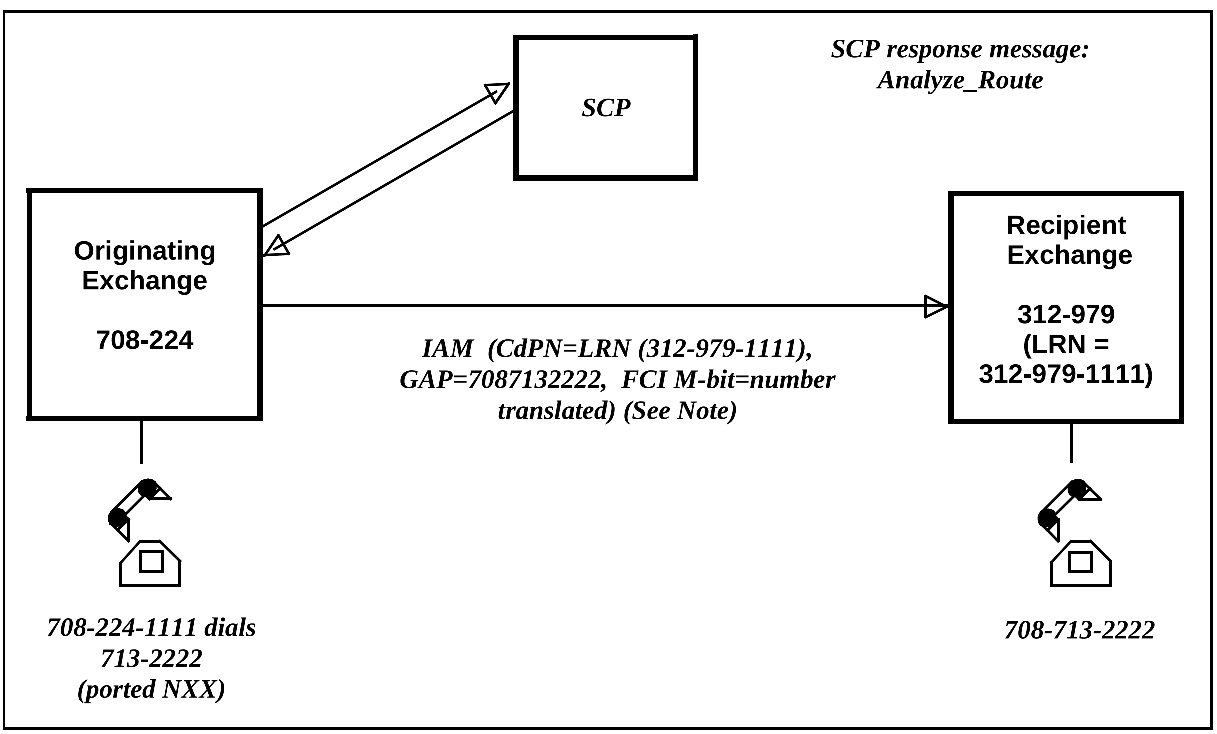
Figure 10-6 illustrates a call placed to a ported DN through a direct ISUP trunk to the recipient exchange. The call proceeds as follows:

**1)** The ported NPA-NXX (708-713) is recognized in the originating exchange's translator. The originating exchange determines that the called digits are not associated with one of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP.

**2)** The LRN returned by the SCP is processed by the originating exchange's translator. The originating exchange's translations for LNP routes the call directly to the recipient exchange.

**3)** The recipient exchange recognizes its LRN in the calling information routed by the originating exchange. The digits in the received GAP are translated by the recipient exchange and the call terminates.

**Figure 10-6: LNP Call Processing - Originating Exchange with Direct ISUP Trunk to Recipient Exchange**




***Note:*** *In this case, the SCP response message contains the GAP and FCI M-bit. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

### Call placed to a non-ported DN through a direct trunk to the recipient exchange

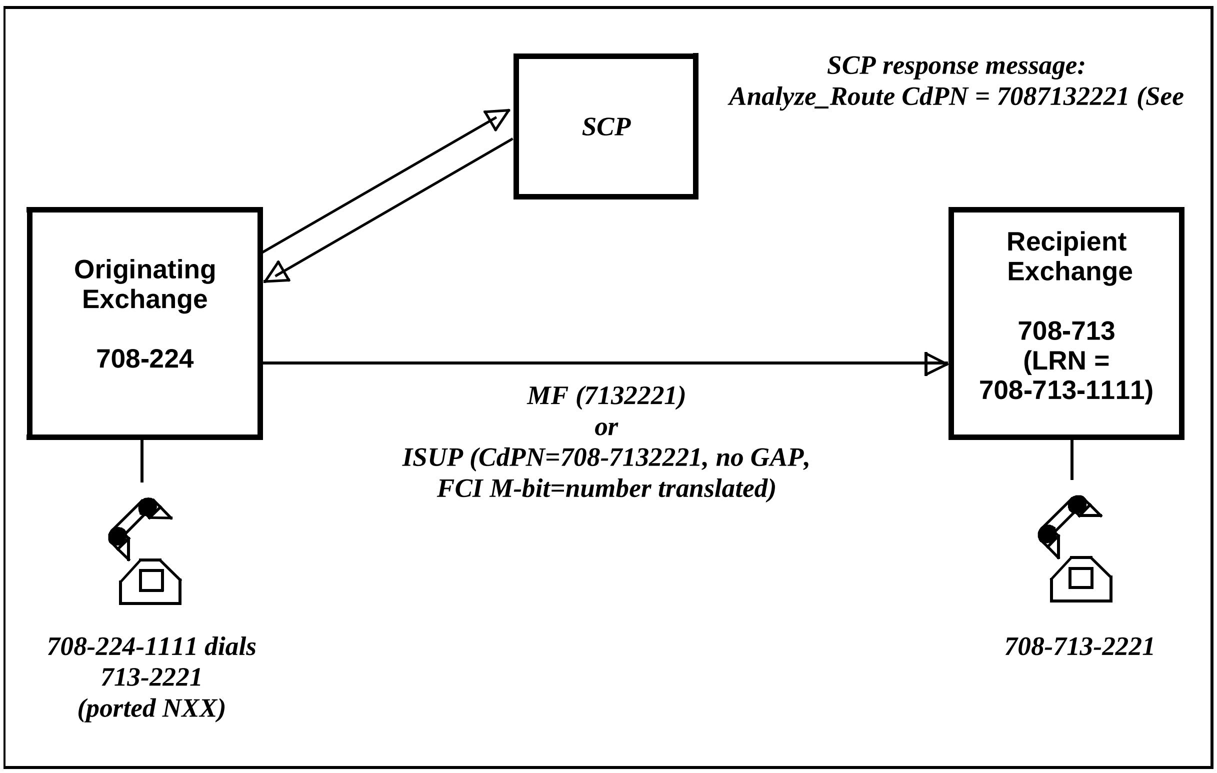
Figure 10-7 illustrates a call placed to a non-ported DN through a direct trunk to the recipient exchange. The call proceeds as follows:

**1)** The ported NPA-NXX (708-713) is recognized in the originating exchange's translator. The originating exchange determines that the called digits are not associated with one of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP.

**2)** The SCP recognizes this as a non-ported DN and returns the called party number. The called party number is then re-translated by the originating exchange and the call is routed to the recipient exchange.

**3)** Translation at the recipient exchange results in call termination.

**Figure 10-7: LNP Call Processing - Originating Exchange with Direct Trunk to Recipient Exchange (non-ported DN)**



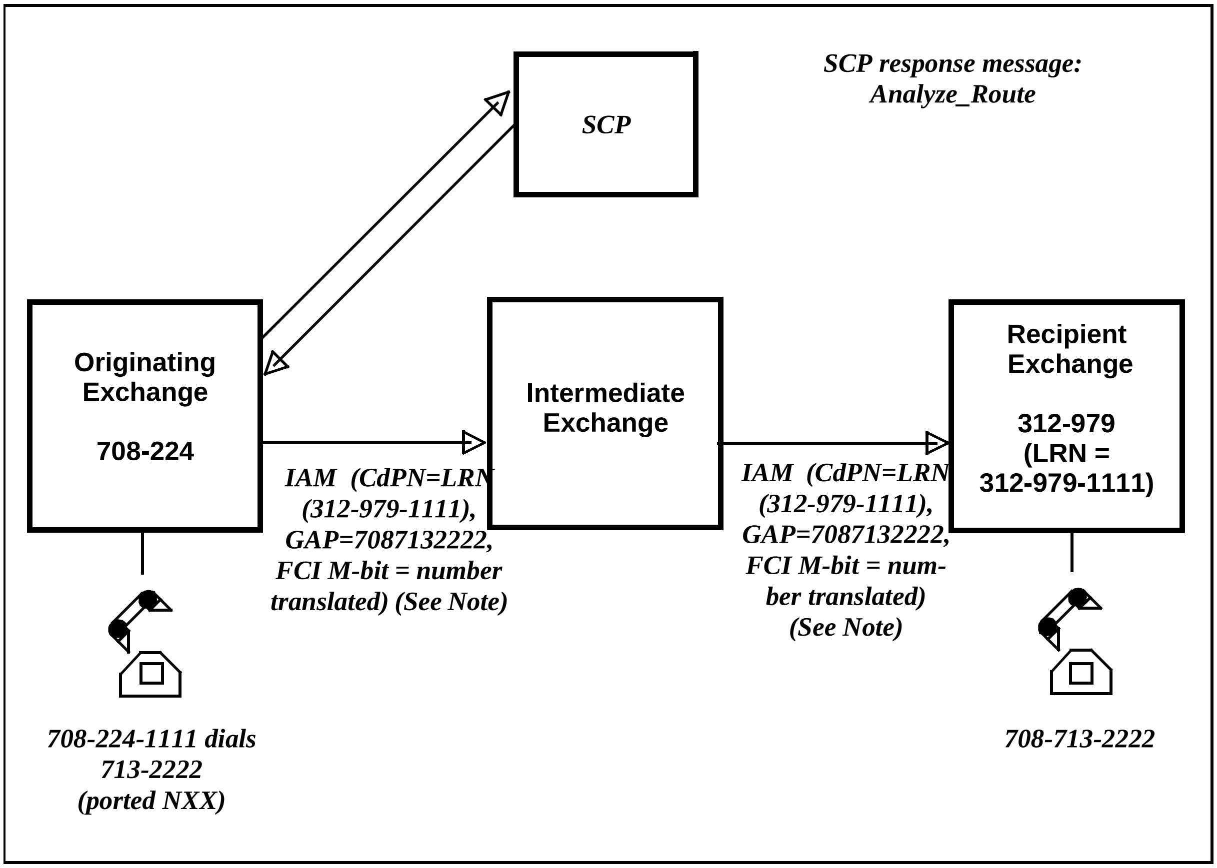

*Note:* *The SCP response message does not contain a GAP or FCI M-bit. These are implied by the response (CdPN not changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

### Call placed indirectly to a recipient exchange

Figure 10-8 illustrates a call placed indirectly to a recipient exchange. The call proceeds as follows:

1) The ported NPA-NXX (708-713) is recognized in the originating exchange's translator. The originating exchange sends a query to the SCP.
2) The SCP returns an LRN and the LRN is then routed to an intermediate exchange.
3) Since the intermediate exchange is not identified by the LRN, subsequent translation routes the call to the recipient exchange.
4) The recipient exchange recognizes the LRN and translates the digits received in the GAP. The call terminates.

**Figure 10-8: LNP Call Processing - Originating Exchange with Indirect Completion to Recipient Exchange**




*Note:* *The SCP response message contains a GAP and FCI M-bit. These are implied by the response (CdPN changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

### Call in which the donor exchange performs an LNP query

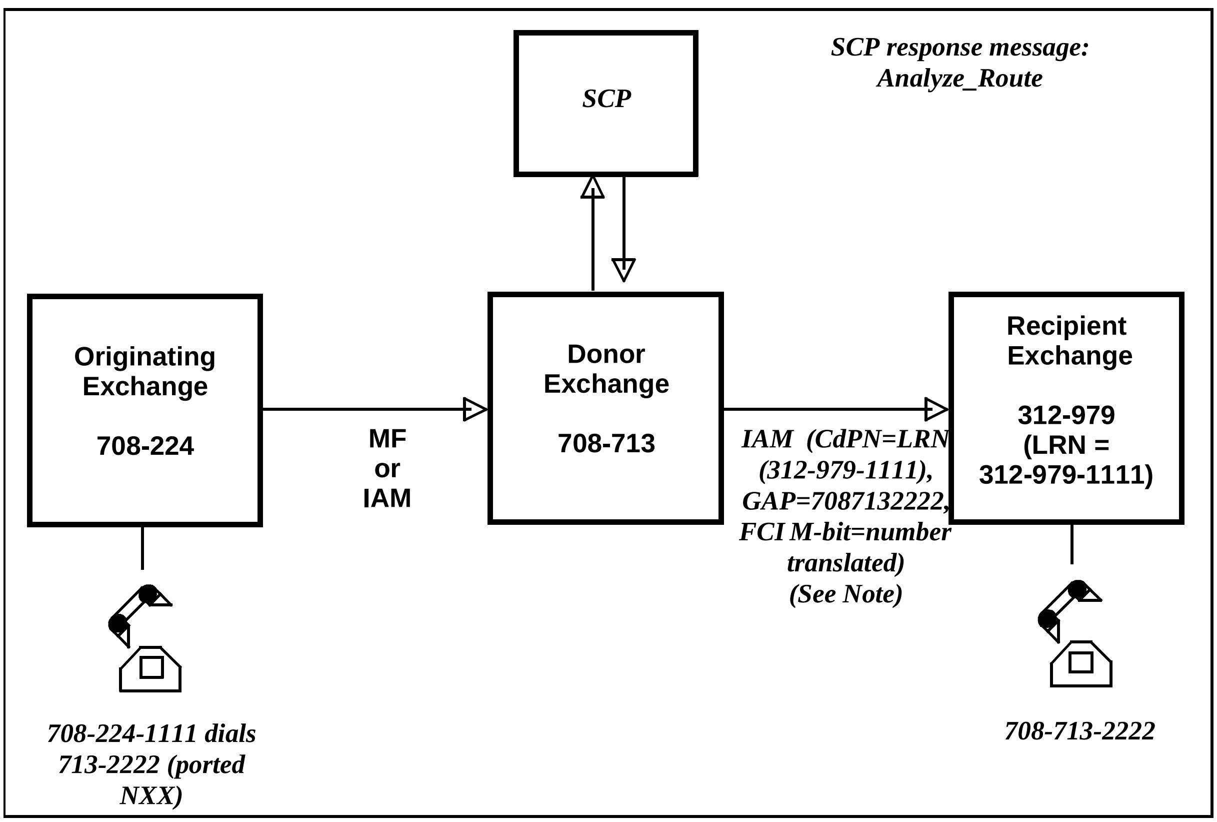
Figure 10-9 illustrates a call in which the donor exchange performs an LNP query. The call proceeds as follows:

**1)** Since the originating exchange does not have SCP access capability, the call is routed (through normal pre-LNP routing) to the donor exchange.

**2)** The donor and recipient exchanges are in the same rate center. The ported NPA-NXX (708-713) is recognized in the donor exchange's translator. The donor exchange determines that the called digits are not associated with any of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP. The LRN returned by the SCP is processed by the donor exchange's translator and the LRN is routed to the recipient exchange.

**3)** The recipient exchange recognizes its LRN in the calling information routed by the originating exchange. The digits in the received GAP are translated by the recipient exchange and the call terminates.

**Figure 10-9: LNP Call Processing - Donor Exchange Performs LNP Query**




*Note:* *The SCP response message contains a GAP and FCI M-bit. These are implied by the response (CdPN changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

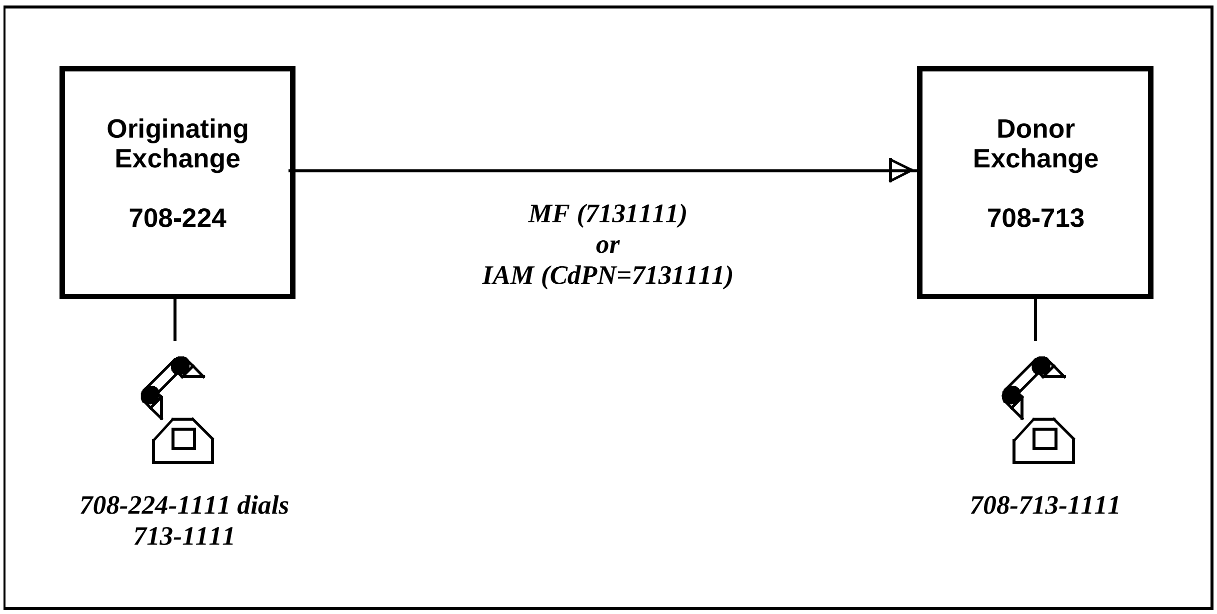
### Call placed to a non-ported DN when the originating exchange cannot access the SCP

Figure 10-10 illustrates a call placed to a non-ported DN when the originating exchange cannot access the SCP. The call proceeds as follows:

**1)** Digit analysis at the originating exchange results in the call being routed to the donor exchange.

**2)** The donor and recipient exchanges are in the same rate center and the NPA-NXX (708-713) is recognized in the donor exchange translator. The donor exchange determines that the called number is associated with one of its subscribers and the call terminates.

**Figure 10-10: LNP Call Processing - Originating Exchange to Donor Exchange with Non-ported DN**



### Call placed to a ported NXX through a direct MF trunk to the recipient exchange

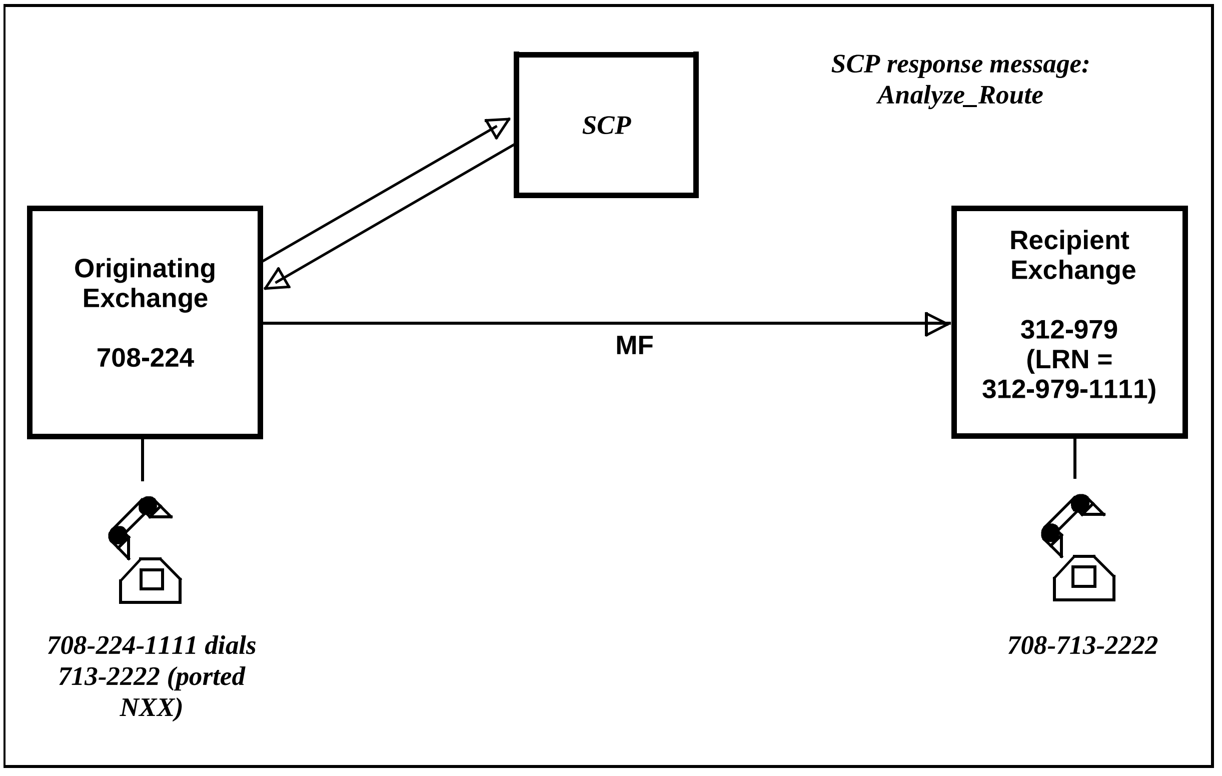
Figure 10-11 illustrates a call placed to a ported NXX through a direct MF trunk to the recipient exchange. The call proceeds as follows:

**1)** The ported NPA-NXX (708-713) is recognized in the originating exchange's translator. The originating exchange sends a query to the SCP, then sends the returned LRN through its LNP translator, and determines an MF trunk group out of the office.

**2)** Using MF signaling, the originating exchange signals the dialed number to the receiving exchange using existing procedures.

**3)** The recipient exchange processes the incoming digits and completes the call.

**Figure 10-11: LNP Call Processing - Originating Exchange with Direct MF Trunk to Recipient Exchange**




*Note:* *The SCP response message contains a GAP and FCI M-bit. These are implied by the response (CdPN changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

### Call in which an intermediate exchange with incoming MF signaling performs an LNP query

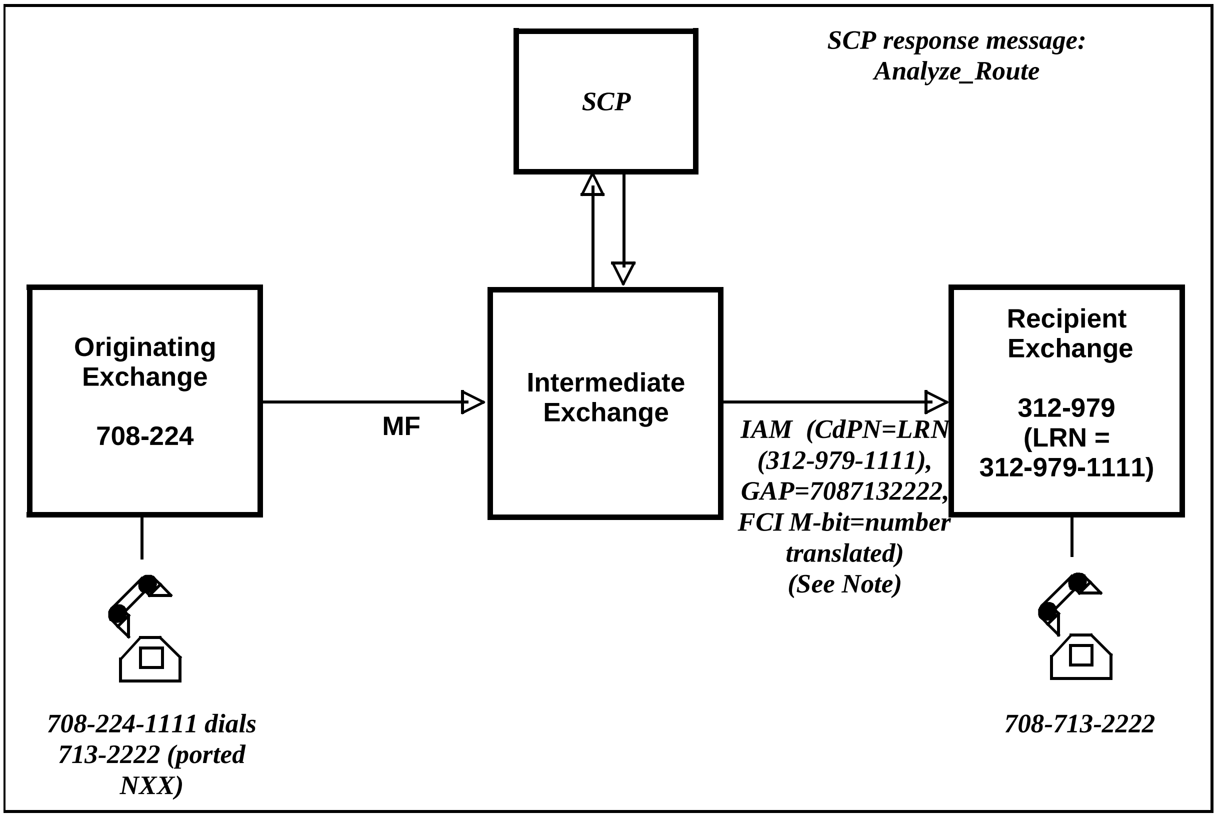
Figure 10-12 illustrates a call in which an intermediate exchange with incoming MF signaling performs an LNP query. The call proceeds as follows:

**1)** Since the originating exchange does not have SCP access capability, the call is routed (through normal pre-LNP routing) to the intermediate exchange.

**2)** The ported NPA-NXX (708-713) is recognized in the intermediate exchange's translator. The intermediate exchange determines that the called digits are not associated with any of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP. The LRN returned by the SCP is processed by the intermediate exchange's translator and the LRN is routed to the recipient exchange.

**3)** The recipient exchange recognizes its LRN in the calling information routed by the originating exchange. The digits in the received GAP are translated by the recipient exchange and the call terminates.

**Figure 10-12: LNP Call Processing - Intermediate Exchange with Incoming MF Signaling Performs LNP Query**




*Note:* *The SCP response message contains a GAP and FCI M-bit. These are implied by the response (CdPN changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

### Call in which an intermediate exchange with outgoing MF signaling performs an LNP query

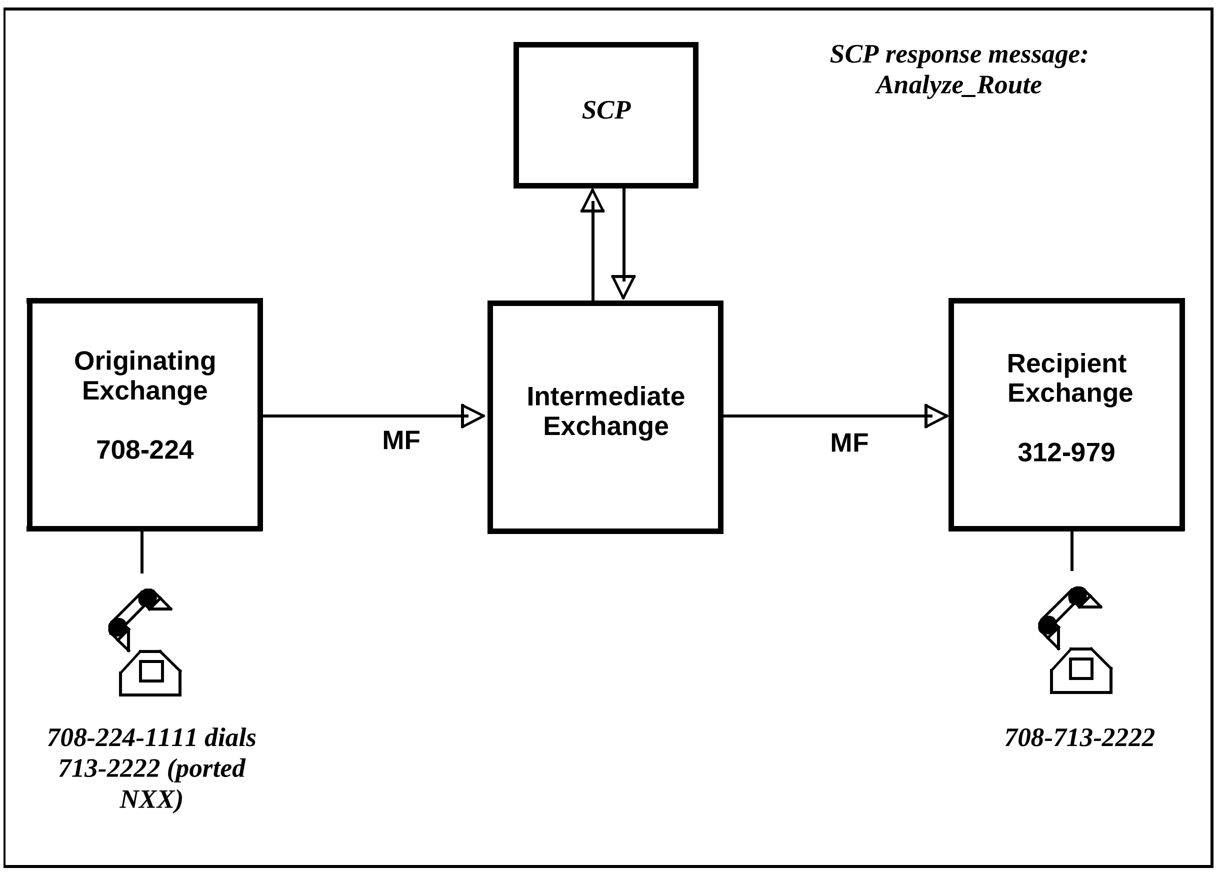
Figure 10-13 illustrates a call in which an intermediate exchange with outgoing MF signaling performs an LNP query. The call proceeds as follows:

**1)** Since the originating exchange does not have SCP access capability, the call is routed (through normal pre-LNP routing) to the intermediate exchange.

**2)** The intermediate and recipient exchanges are in the same rate center. The ported NPA-NXX (708-713) is recognized in the intermediate exchange's translator. The intermediate exchange determines that the called digits are not associated with any of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP. The LRN returned by the SCP is processed by the intermediate exchange's translator and the LRN is routed to the recipient exchange. The dialed digits are outpulsed.

**3)** The digits received are translated by the recipient exchange and the call terminates.

**Figure 10-13: LNP Call Processing - Intermediate Exchange with Outgoing MF Signaling Performs LNP Query**



### Call in which an intermediate exchange with outgoing MF/ISUP signaling performs an LNP query

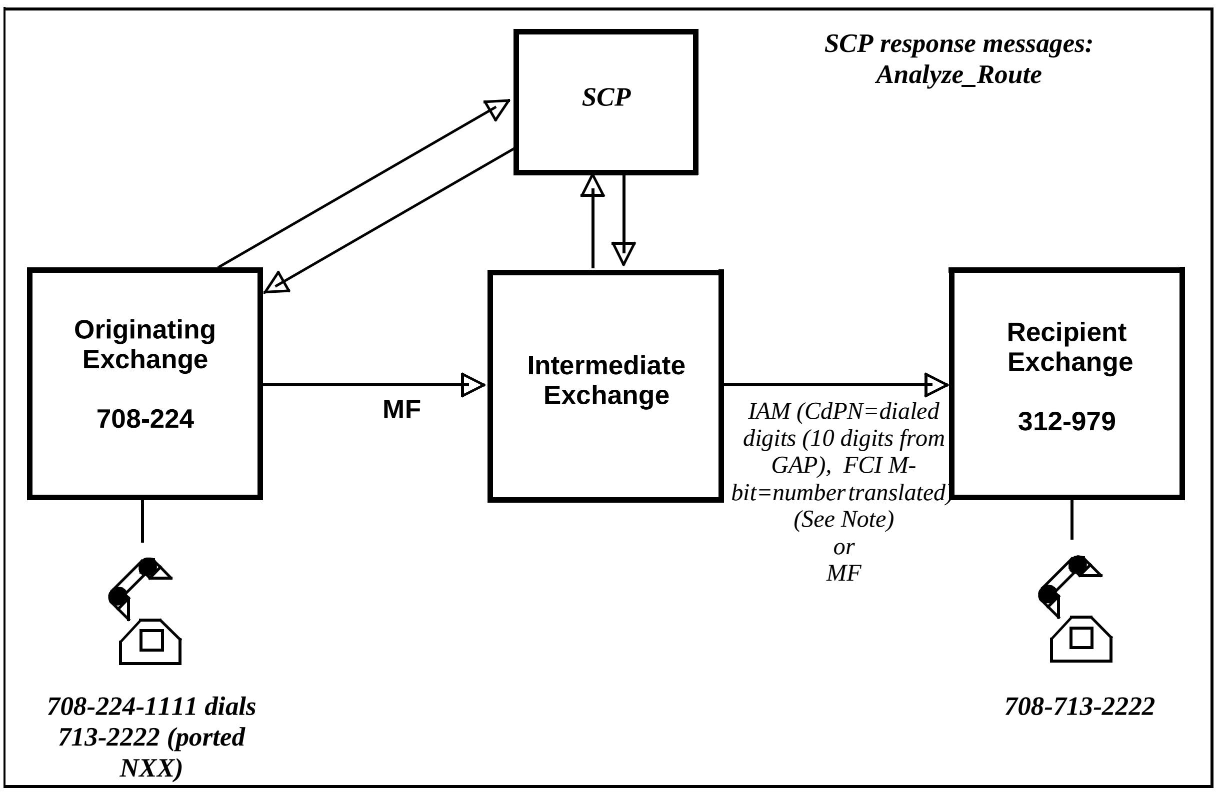
Figure 10-14 illustrates a call in which an intermediate exchange with outgoing MF/ ISUP signaling performs an LNP query. The call proceeds as follows:

**1)** After an SCP query and LNP routing, the originating exchange routes the call to the intermediate exchange (outpulsing MF dialed digits).

**2)** The ported NPA-NXX (708-713) is recognized in the intermediate exchange's translator. The intermediate exchange determines that the called digits are not associated with any of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP. The LRN returned by the SCP is processed by the intermediate exchange's translator. Since the signal ported number trunk group option is indicated, the dialed digits are to be outpulsed instead of the LRN. The LRN is routed to the recipient exchange (outpulsing the MF dialed digits, or an IAM containing dialed digits).

**3)** The called digits are translated by the recipient exchange and the call terminates.

**Figure 10-14: LNP Call Processing - Intermediate Exchange with Outgoing MF/ISUP Signaling Performs LNP Query**




*Note:* *The SCP response message contains a GAP and FCI M-bit. These are implied by the response (CdPN changed) and are associated with the call following the response. The GAP and FCI M-bit are stored in the outgoing IAM if ISUP is used.*

## Foreign LRN

If a switch in the area of portability supports multiple local call areas, and if the LRN of that switch is in another call area, calls between that switch and the local switch may be rated as toll calls. Translations rates the call as it was dialed and does not re-rate it when the LRN is translated. To allow this type of non-toll call to complete, a TOL N leg and a non-toll route between the switches must exist in translations.

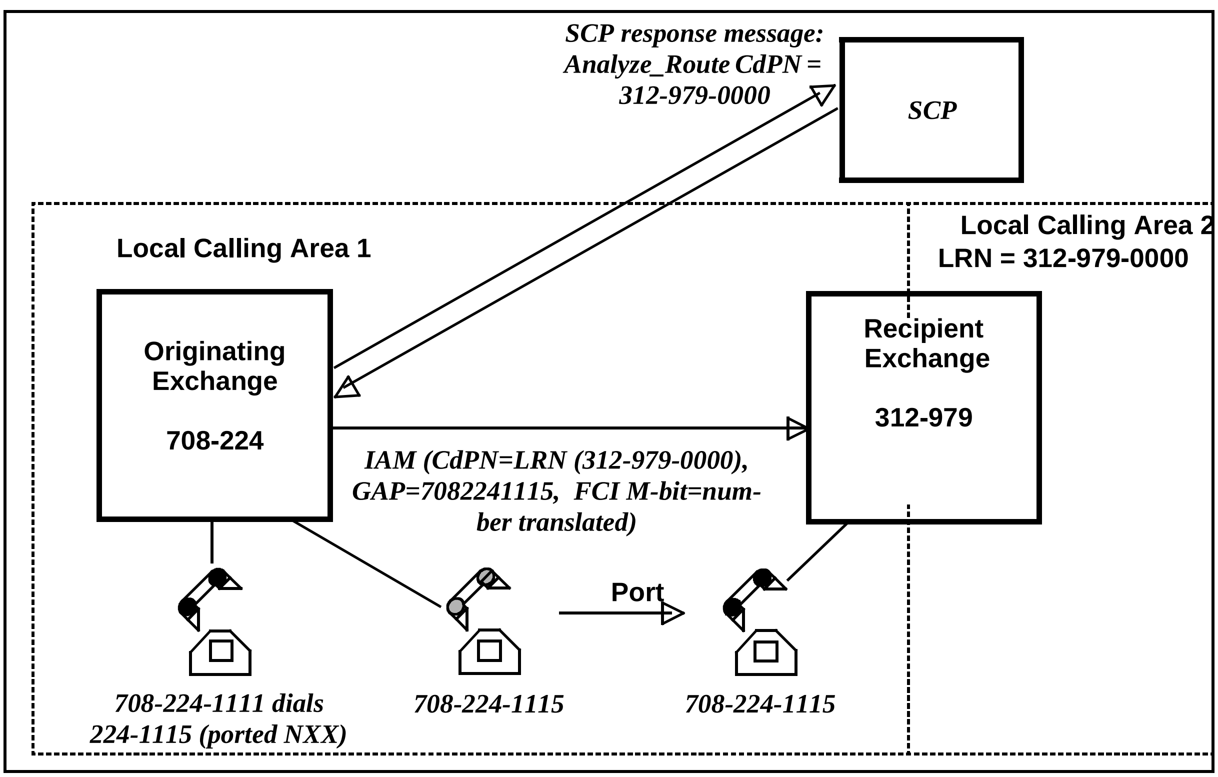
Figure 10-15 illustrates a call placed to a ported DN through a direct ISUP trunk to the recipient exchange. The call proceeds as follows:

**1)** The ported NPA-NXX (708-224) is recognized in the originating exchange's translator. The originating exchange determines that the called digits are not associated with one of its subscribers and that an LNP trigger exists for the dialed DN, and then sends a query to the SCP. The call is rated as non-toll.

**2)** The LRN returned by the SCP is processed by the originating exchange's translator. Translations routes the call directly to the recipient exchange but does not re-rate the call when the LRN is translated.

**3)** The recipient exchange recognizes its LRN in the calling information routed by the originating exchange. The digits in the received GAP are translated by the recipient exchange and the call terminates.

**Figure 10-15: LNP Call Processing - Foreign LRN**



## Porting an LNP number

The CCS7 software package is a prerequisite for using TCAP messaging between the DMS-10 SSP and the SCP.

### Porting a number

To port a number, the following steps are performed:

**1)** The number to be ported (at the donor switch) must first be disassociated (that is, unassigned) from its current physical equipment. An LNP trigger must then be associated with the ported number's 6 through 10-digit trigger (NPA-NXX). The operating company, in coordination with a third-party agency responsible for administering LNP numbers, verifies that the ported number is available for assignment. The donor switch's translations for LNP is then updated to route to the LRN of the recipient exchange.

**2)** A new office code for the ported number is then built in the recipient switch (the switch that the number is being ported to). Line equipment is then assigned to the ported number.

**3)** An LNP trigger must be associated with the ported number's NPA-NXX in all LNP-capable switches (switches with access to the SCP) in the rate center containing the ported number. The translations for LNP in each of these switches must be updated to route the LRN to the recipient exchange.

**4)** The SCP is then updated to associate the ported DN with the LRN of the recipient switch.

### Deleting a ported number

After a subscriber has changed to a different service provider (the subscriber's DN has been ported to another switch) and subsequently discontinues service, the DN is reestablished on the donor (original) switch as outlined in the following steps:

**1)** The SCP is updated to disassociate the ported DN from the LRN of the former recipient switch.

**2)** The third-party agency responsible for administering LNP numbers determines whether the DN is the last within the NPA-NXX. If it is, all LNP-capable switches in the rate center of the ported DN should remove the LNP trigger associated with the ported number's NPA-NXX.

**3)** The recipient switch either removes the office code associated with the ported DN or updates translations such that calls placed to the office code will be routed to the original donor switch. The DN is disassociated from the switch's line equipment.

**4)** The donor switch removes the LNP trigger from the NPA-NXX; calls placed to the number will then be given intercept treatment. The DN is then available at the donor switch for reassignment.

## Number Pooling

Directory numbers are currently allocated by extending ownership of an entire NXX group (for example, 433-0000 through 433-9999) to a single central office. Many central offices, however, do not use all of these allocated DNs. To improve utilization of directory numbers, Number Pooling enables allocation of directory numbers to a shared reservoir associated within a geographic area so that blocks of directory numbers smaller than an entire NXX code can be assigned to different service providers. Thus, instead of an office being assigned a block of 10,000 directory numbers (NPA-NXX), the office can be assigned a block of 1000 directory numbers (NPA-NXX-X).

### Number allocation

Existing network routing mechanisms route calls to a specific office and switch according to the first six digits of the called number (NPA-NXX). Number Pooling ignores the association of an NXX with an office and relies on Local Number Portability (LNP) processing for call routing. Thus, Number Pooling can operate only between LNP-capable switches. Number Pooling also can operate only between switches in the same rate center.

To configure Number Pooling, an NXX is assigned to a service provider's switch, called a “code holder,” and blocks consisting each of 1000 DNs are assigned to service provider switches called “block holders.” The SCP LNP database associates the DNs with a Location Routing Number (LRN) that identifies a particular switch in the LNP network that serves the DNs.

### Interswitch Call to Pooled DN

Figure 10-16 illustrates basic interswitch pooled DN call flow:

**1)** A block of DNs is pooled from Switch X to Switch Y. Switch Y is the block holder for 630-224-2. A subscriber on Switch Y is assigned the pooled DN 630-224-2001.

**2)** The subscriber at 630-224-1001 (Switch X) dials the subscriber at 630-224-2001 (Switch Y). Switch X determines that the called digits are not associated with one of its subscribers and then sends a query to the SCP.

**3)** The LRN returned by the SCP is processed by Switch X's translator. Switch X's translations routes the call directly to Switch Y, which translates the digits, and the call terminates.

### Intraswitch Call to Pooled DN

Figure 10-17 illustrates basic intraswitch pooled DN call flow:

**1)** A block of DNs is pooled from Switch Y to Switch X. Switch X is the block holder for 630-979-9. A subscriber on Switch X is assigned the pooled DN 630-979-9000.

**2)** The subscriber at 630-224-1001 dials the subscriber at 630-979-9000. Switch X terminates the call to 630-979-9000.

**Figure 10-16: Number Pooling Call Processing - Interswitch Call to Pooled DN**

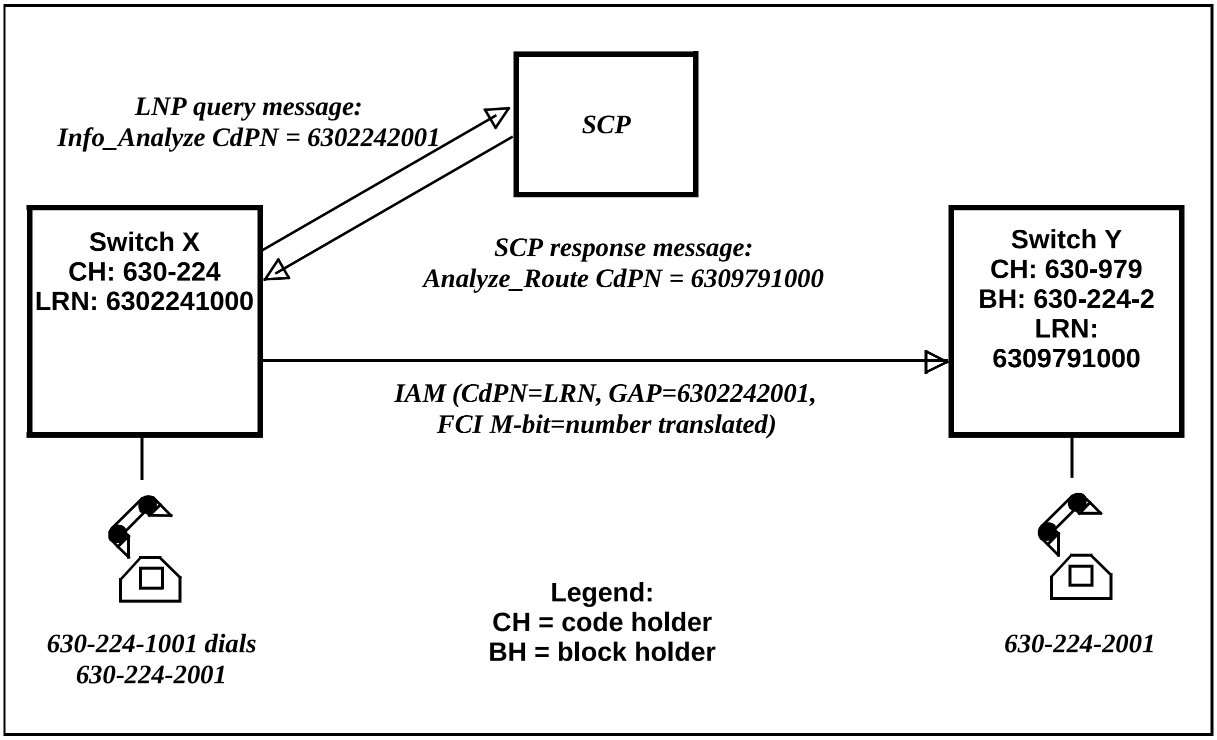
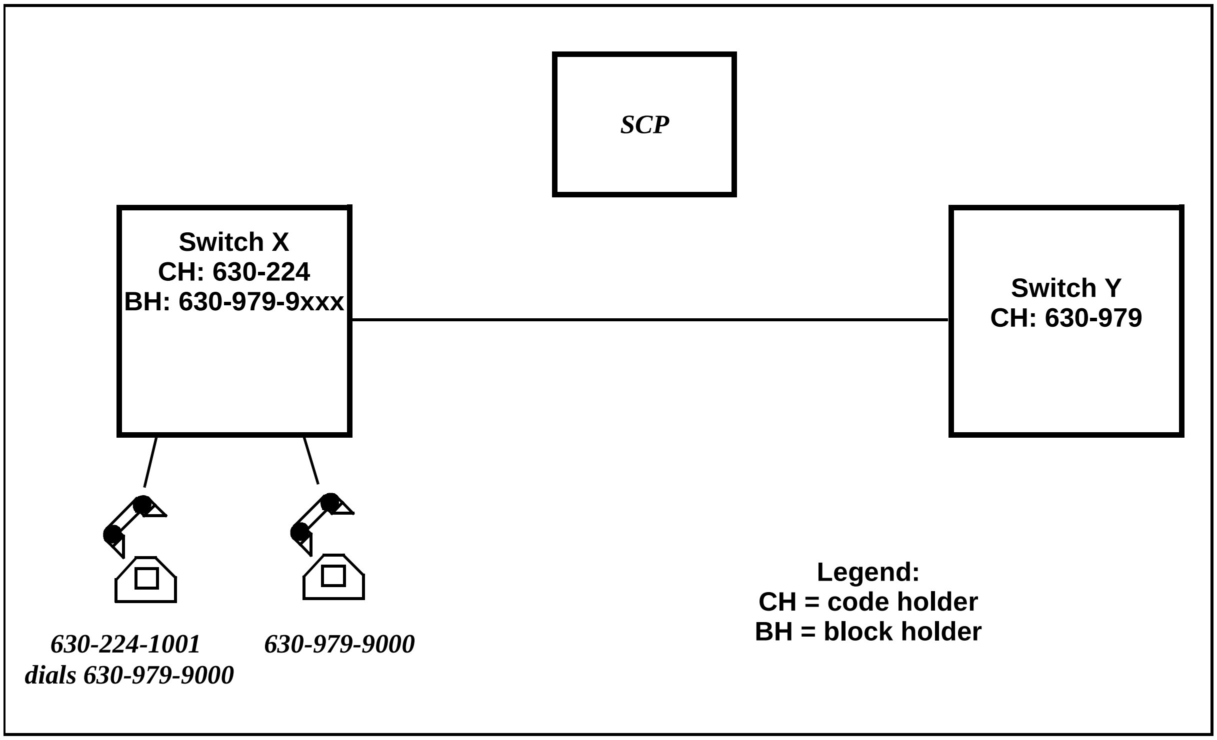


**Figure 10-17: Number Pooling Call Processing - Intraswitch Call to Pooled DN**



### NP-reserved (NPR) marking and generic condition

When a block is pooled, the SCP database is updated with the LRN of the recipient switch for the DNs in the pooled block. Thus, routing to DNs in this block that are unassigned to subscribers may occur. Since such calls are not mis-routed, the ISUP release message cause code 26, "misrouted call to a ported number," normally returned when an LRN-routed call attempts to terminate to an unassigned number, is inaccurate. To prevent cause code 26 from being sent for such calls, unassigned, pooled DNs can be marked as "NP-reserved." An "NP-reserved" generic condition then provides treatment for calls routed to unassigned, NP-reserved DNs.

NP-reserved marking and generic condition are applied in the following situations:

> ***Note:*** *"Vacant DNs" are DNs assigned the Vacant DN generic condition.*

- The marking is automatically applied to vacant DNs when a new thousands group is created and designated as a pooled group - that is, the NP-reserved marking is not applied to those DNs that are not vacant.
- The marking is automatically applied to vacant DNs when a thousands group is changed and designated as a pooled group.
- The marking can be manually applied by operating company personnel, using prompting sequence DN (ICP) (see 297-3601-311, *Data Modification Manual*), to intercept DNs to the "NPR" generic condition. A pooled DN that is deleted is assigned the DNIC generic condition. It is the responsibility of the operating company to assign the NP-reserved marking after snapback.

NP-reserved marking is removed in the following situations:

- The marking is automatically removed when a thousands group is changed to indicate that the thousands group is no longer a pooled group. All DNs that are marked NP-reserved are also marked vacant. Any DNs not marked NP-reserved remain at their current assignment.
- The marking can be manually removed by operating company personnel, using the DN (ICP) prompting sequence (see 297-3601-311, *Data Modification Manual*), to either intercept DNs to a generic condition other than "NPR," to a route, or to a screen.

### Ported-in (PRTI) marking and generic condition

When a DN that does not belong to a native thousands group is ported in to a switch, the telco declares a new thousands group. To ensure that the Telco knows that the DNs in the thousands group, except for the ported-in DN, are unavailable for assignment, a “ported-in” generic condition can be declared. Thus, all DNs in the thousands group can be intercepted, marked as ported-in (PRTI), and provided with appropriate treatment for this generic condition. The generic condition applies only to thousands groups that are not pooled.

The PRTI marking and generic condition are applied in the following situations:

- The PRTI marking is automatically applied to vacant DNs when a new thousands group is created and designated as a ported-in group.
- The PRTI marking is automatically applied to vacant DNs when a thousands group is changed and designated as a ported-in group - that is, the PRTI marking is not applied to those DNs that are not marked vacant.
- The PRTI marking can be manually applied by operating company personnel to a single DN or to a range of DNs belonging to a ported-in thousands group, using the DN (ICP) prompting sequence (see 297-3601-311, *Data Modification Manual*), by intercepting the DNs to the “PRTI” generic condition.

The PRTI marking and generic condition are removed in the following situations:

- The marking is automatically removed when a thousands group is changed to indicate that it is no longer a ported-in group. All DNs that are marked PRTI are marked vacant. Any DNs not marked PRTI remain at their current assignment.
- The marking can be manually removed by operating company personnel, using the DN (ICP) prompting sequence (see 297-3601-311, *Data Modification Manual*), by intercepting the DNs to a generic condition other than PRTI, to a route, or to a screen.

### Ported-out marking

A “ported-out” marking is applied to pooled DNs that are ported to another switch. The ISUP release message cause code 26, “misrouted call to a ported number,” is returned when a call is mis-routed to a DN marked ported-out. For example, if a station on switch A places a call to a pooled DN on switch B that has been ported to switch C, and switch A's database still shows the called DN associated with the LRN of switch B, switch B will provide cause code 26 treatment for the call attempt.

The “ported-out” marking is not automatically applied. It can be manually applied to a DN by intercepting the DN to the “LNP” generic condition.

### Code holder pooled block marking

A “code holder pooled block” marking is applied to pooled DNs that are pooled out of the code holder switch to the block holder switch and, thus, are unavailable for assignment. The marking aids the operating company in identifying available DNs. The marking should be applied at the code holder switch when a block of DNs is pooled to another switch and the code holder does not remove the thousands group for the pooled block. This marking is applicable only to the code holder switch.

The “code holder pooled block” marking is not automatically applied. It can be manually applied to a DN by intercepting the DN to the “CHPB” generic condition.

### Restrictions and Limitations

Prior to the introduction of the Number Pooling feature, an individual DN, or a sub-range of DNs, could not be intercepted to another generic condition or route when the DNs fell within a range that was already intercepted at a higher level. For example, intercepting a hundreds group would preclude intercepting the tens group to another generic condition or route. With the introduction of the Number Pooling feature, this requirement has been relaxed to allow an individual DN or range of DNs (thousands, hundreds, and tens group) to be intercepted when a higher-level interception has already taken place.

A limitation occurs, however, when a range (thousands, hundreds, or tens level) is intercepted to a non-Vacant DN generic condition and the sub-range is intercepted to the Vacant DN generic condition. If a dump and sysload were to occur, the sub-range that was marked Vacant DN would not reload as Vacant DN but would share the same assignment as the higher-level intercept. For example, if the hundreds group of 921 11 is intercepted to NP-Reserved and the sub-range in the tens group of 921 110 is intercepted to Vacant DN, a dump and sysload would result in the tens group being intercepted to NP-Reserved instead of Vacant DN. This anomaly occurs only when a sub-range is intercepted to Vacant DN while a higher-level intercept exists.

## Duplicate NXX feature

The Duplicate NXX feature enables the DMS-10 switch to support subscriber numbers that have common thousands groups but different HNPAs.  This capability is necessary to support subscriber numbers being ported into an LNP-capable network from an adjacent network that has the same thousands group but different HNPA. As shown in Figure 10-18 for example, a number with HNPA 214 and thousands group 718-7 can be ported from switch A to switch B, which already has thousands group 718-7 but with a different HNPA, 972.

Prior to the introduction of this feature, directory numbers could be defined on the DMS-10 switch only as 7-digit numbers with implied HNPAs. The Duplicate NXX feature now enables directory numbers to be defined as ten-digit numbers. This prevents any ambiguity that might arise during call processing involving DNs with common thousands groups but different HNPAs on the switch. Thus, when placing a call to such a number on the switch from a different HNPA, the subscriber must dial ten digits; otherwise the switch attempts to place the call to a DN with an HNPA associated with the subscriber's directory number. If the switch is unable to determine the HNPA of the calling subscriber, the thousands group of the calling party is used to resolve the ambiguity. If the subscriber dials a ten-digit DN that translates to a thousands group not found on the switch, translations checks for an HNPA split. If the HNPA is being split, and if the permissive dialing period is still active, the mate HNPA (assigned in Overlay AREA) is used to determine the thousands group to be used. Sample dialing scenarios are shown in Figure 10-19.

**Figure 10-18: Duplicate NXXs after porting**

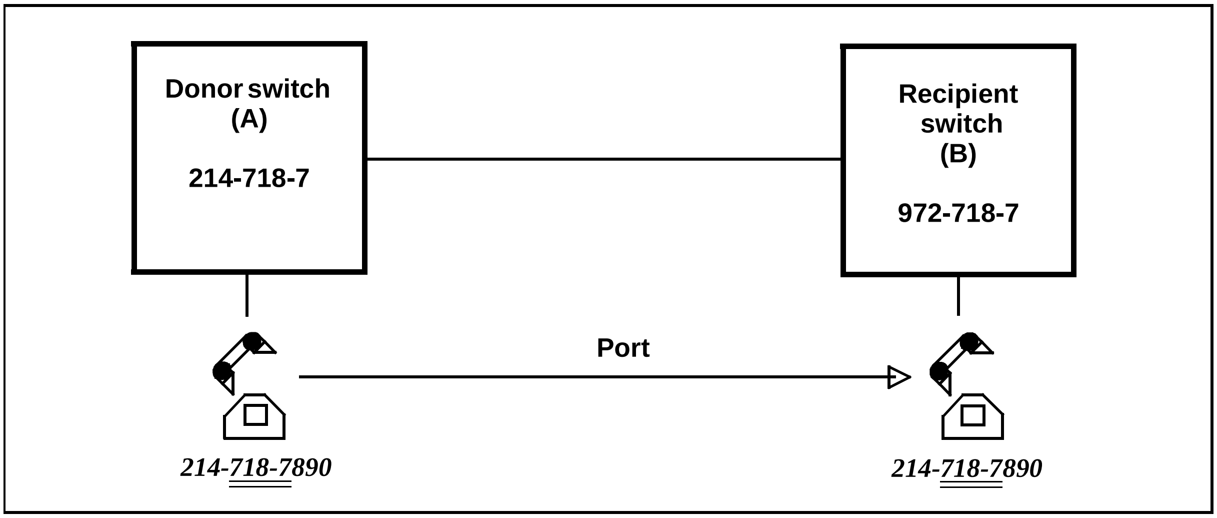
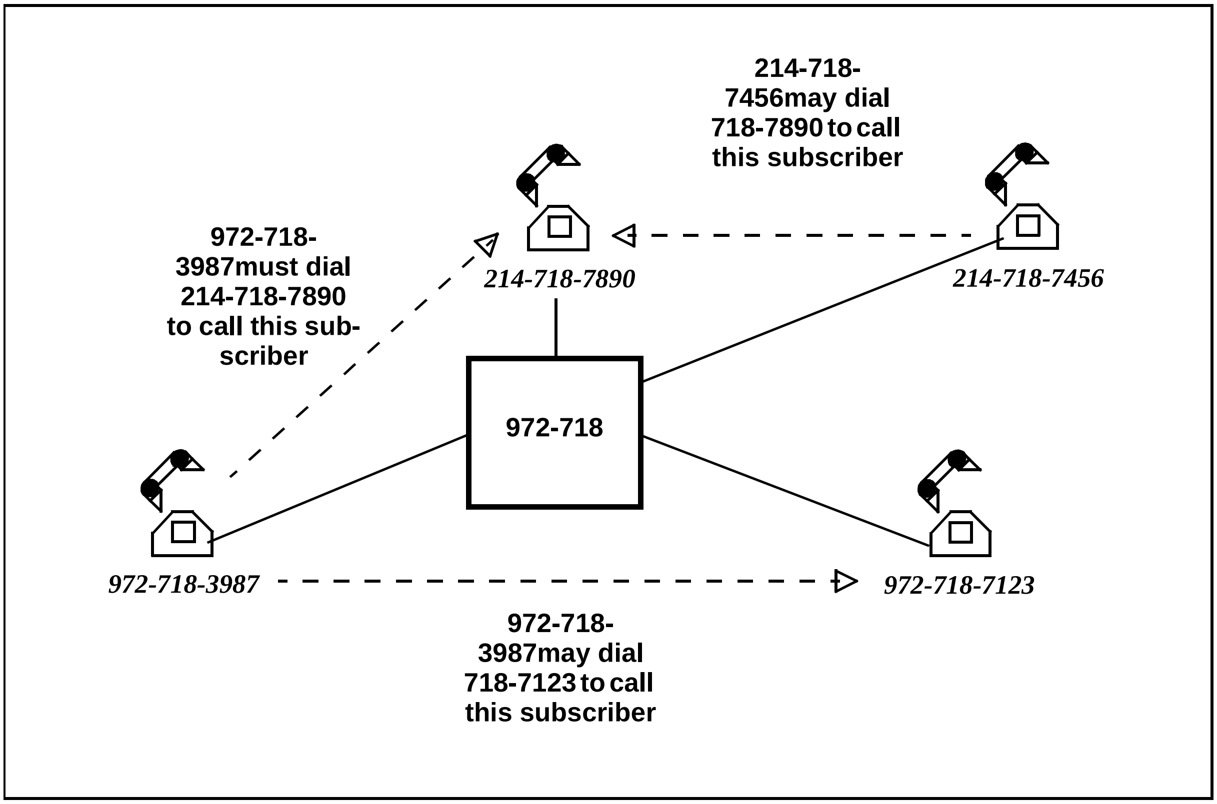


**Figure 10-19: Duplicate NXX dialing examples**



Although the Duplicate NXX feature enables ten-digit DNs to be defined on the switch, only seven digits need to be entered by operating company personnel if only one HNPA is associated with the thousands group of the DN; a ten-digit number must be entered, however, if the thousands group of the DN is associated with multiple HNPAs. The number of HNPAs that can be defined on the switch is 32.

The following conditions pertain to the Duplicate NXX feature:

- DNs assigned to an OPM or RLCM using the NTMX45 processor pack with the feature E10D activated, support up to 8 HNPA(s) with three-to-ten digit dialing in ESA mode. All other remotes must assign DNs to the same HNPA. The ESA translators for these remotes are not able to resolve thousands group ambiguity.
- All DNs on eight- or ten-party lines, or on Operator Number Identification multiparty lines, must be in the same HNPA, because billing for each number is determined using the HNPA associated with the subscriber's line on the interface with the first ring code.
- Seven digit dialing across HNPA boundaries is still allowed. Ten-digit dialing is required, however, if the HNPA for a called thousands group is ambiguous.
- CAMA CO screening does not check the HNPA in duplicate NXX numbers.

# Section 11: CALEA

## Introduction

U.S. Public Law 103-414, the Communications Assistance for Law Enforcement Act (CALEA) of 1994 was enacted to amend Title 18, United States Code, to make clear a telecommunications carrier's duty to cooperate in the interception of communications for law enforcement purposes. CALEA mandated that the U.S. Attorney General implement the provisions of the law. The Attorney General, in turn, designated the Federal Bureau of Investigation (FBI) to implement CALEA on behalf of all federal, state, and local law enforcement agencies, while maintaining a balance of public safety, the public's right to privacy, and the industry's ability to remain competitive.

CALEA requires telecommunications equipment manufacturers to provide the technical capability to support lawfully authorized electronic surveillance. From the telecommunications equipment manufacturers' perspective, electronic surveillance refers to the capability to isolate and intercept all wireline, wireless, and electronic communications, to deliver call content and call identifying information to an authorized Law Enforcement Agency (LEA) for a given switch-based subject, and to perform the interception unobtrusively and with minimal interference. Electronic surveillance also refers to the capacity for simultaneous interceptions a switch must be capable of performing during a 24-hour period.

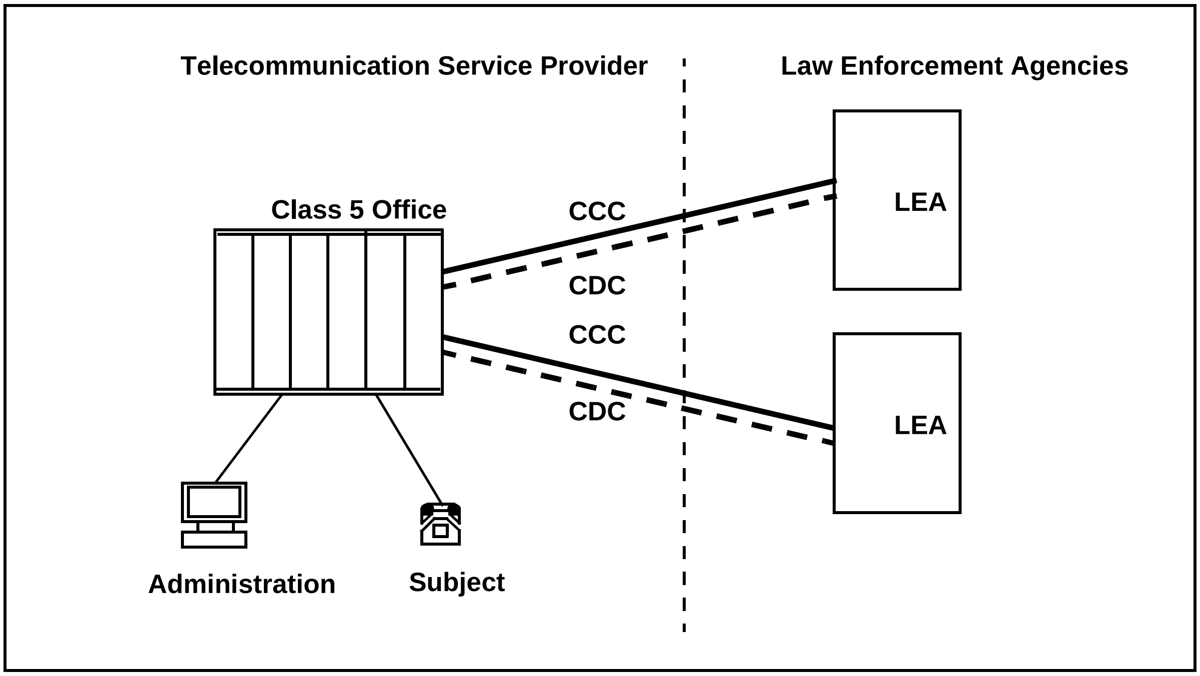
The Communications Assistance for Law Enforcement Agencies (CALEA) feature on the DMS-10 switch provides the functionality, capability, and capacity required to ensure compliance with these requirements.

> ***Note:*** *The DMS-10 switch does not support wireless communication.*

## CALEA Functional Overview

CALEA allows circuit-switched and packet-switched communications of a given switch-based subject to be accessed and delivered to an authorized LEA. Figure 11-1 depicts the physical architecture used by CALEA.

**Figure 11-1: CALEA Network**



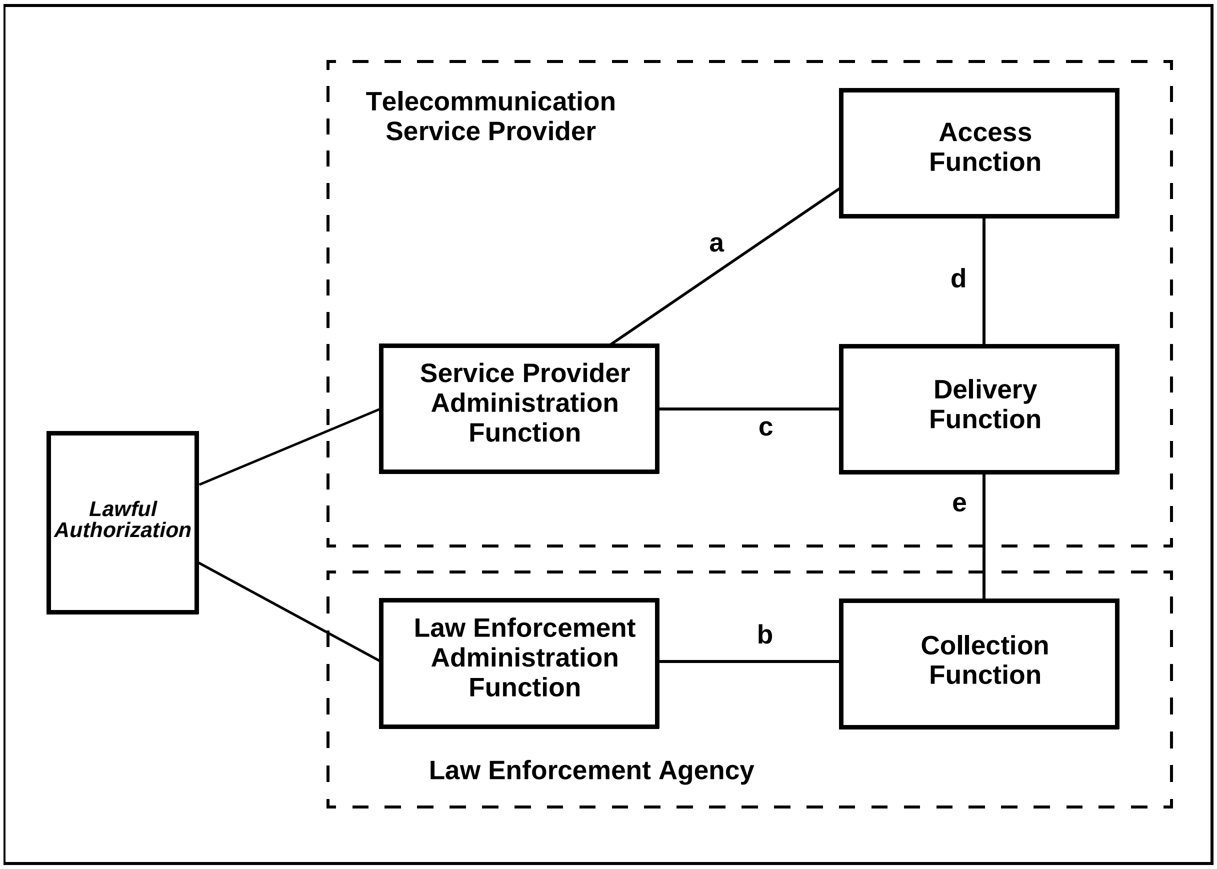

For a given surveillance, call content (for example, circuit-switched data) is delivered over one or more dedicated call content channels (CCC) that are connected to the LEA monitoring center upon surveillance provisioning. These dedicated CCC circuits normally remain connected throughout the duration of the surveillance.

Call identifying information (that is, dialing or signaling information that identifies the origin, direction, destination, or termination of a call) is formatted into discrete messages. The messages are transported to an LEA over a call data channel (CDC), a data connection that connects the switch to the LEA monitoring center. During various states of a monitored call, messages related to the call's progress are delivered over the CDC to the LEA. Each time that a CCC is connected to deliver call content, a message indicating the channel identity is delivered over the CDC.

A subject switch can deliver call content and call data to multiple LEAs at the same time. Surveillance administration is handled by the service provider (that is, the operating company) through the service provider administration function (SPAF). The service provider sends messages to the subject switch to control the set up, activation, modification, and deactivation of surveillance.

There are five broad functions performed by the CALEA feature: access, delivery, collection, service provider administration, and law enforcement administration. The relationships and interfaces between these functions are shown in Figure 11-2.

**Figure 11-2: CALEA Functions**




The *access function* is responsible for isolating a subject's communications, call-identifying information, or both, through use of *intercept access points.*

The *delivery function* is responsible for delivering the intercepted communications to one or more collection functions over the interface designated by the letter *e*. This interface consists of two distinct types of channels: CCCs for call content delivery and CDCs for call data delivery. The interface designated by letter *d* connects the access and delivery functions. The delivery function may either be integrated into the switch or it may be an external device.

The *service provider administration function* is responsible for controlling the access and delivery functions over the interfaces designated by the letters *a* and *c*, respectively.

The *collection function* is responsible for collecting and analyzing the intercepted communication from the subject switch. The law enforcement administration function is responsible for controlling the collection function over the interface designated by letter *b*.

In the DMS-10 switch, an integrated a+c administration interface is provided. Since the delivery function is integrated into the DMS-10 switch, the interface designated by letter *d* is not applicable.

## Call Monitoring

The CALEA feature supports the provisioning of surveillance on a per-directory number basis. Surveillance can be provisioned to any line with Automatic Number Identification (ANI) capabilities. Surveillance of hunt groups and Multiple Appearance Directory Number (MADN) groups is also supported.

Hunt group subjects are uniquely identified within the subject switch by their DN. A hunt group permits a search for an idle directory number within the group in order to complete an incoming call. The CALEA feature isolates monitoring to the specified group member.

MADN subjects are uniquely identified within the subject switch by their DN. A surveillance for a MADN DN monitors all appearances for that DN (that is, the entire MADN group).

A line without ANI capability (that is, an ONI line), a Private Branch Exchange (PBX) or cellular mobile carrier (CELL) line trunk, or an ISDN Primary Rate Interface (PRI) can not be the subject of surveillance.

### Monitoring of subjects, associates, and MRPs

Certain features, such as call forwarding or call transfer, enable the subject's calls to be redirected to another party. When a subject's calls are redirected, they may still be monitored. The party being monitored in this case is called a “monitored replacement party (MRP).” The parties involved in a call with either a subject or an MRP are known as “associates.”

The CALEA feature monitors the subject's service, but not the associates' or MRPs' service. The CALEA feature reports only call events directly affecting the subject's call leg. Call events initiated by an associate or by an MRP, other than the answer or release of the monitored call, are not reported by the CALEA feature. For example, if subject A is involved in a line-to-line call with associate B, and associate B receives a call waiting call from party C, no data will be recorded relating to associate B flashing to answer the call from party C other than the release of the monitored call between subject A and associate B.

The CALEA feature can be used to monitor different subjects and to deliver information to different LEAs. A surveillance is used to monitor one subject and to provide the appropriate information to one LEA. A subject under surveillance by multiple LEAs requires multiple surveillances (that is, each surveillance is independent).

## Communications Access

Circuit-switched and packet-switched call content and call data are accessed internally at the subject switch, through Intercept Access Points (IAP).

Call-identifying information IAPs provide access to information for calls made by an intercept subject or for calls made to an intercept subject. These calls include abandoned calls, incomplete calls, and calls that are redirected (for example, diverted, forwarded, or deflected) by the intercept subject's equipment, facilities, or services. The call-identifying information for the intercept subject is accessed unobtrusively by the IAP. Access to this information does not deny availability of any service either to the subject or to the associates.

The IAPs for content surveillance are used to access the communications of an intercept subject. The content extracted by a content surveillance IAP is delivered over a call content delivery channel (CCC). There are two categories of IAP for content surveillance, circuit IAPs and packet data IAPs.

### Circuit IAP

The circuit IAP, located in the switching network, accesses the call content of circuit-mode communications at the transmit path from the intercept subject and from the associate. This may include calls to an intercept subject that are redirected to another party, or multi-party circuit-mode communication (for example, three-way call).

**Figure 11-3: CALEA Circuit-switched Access Model**

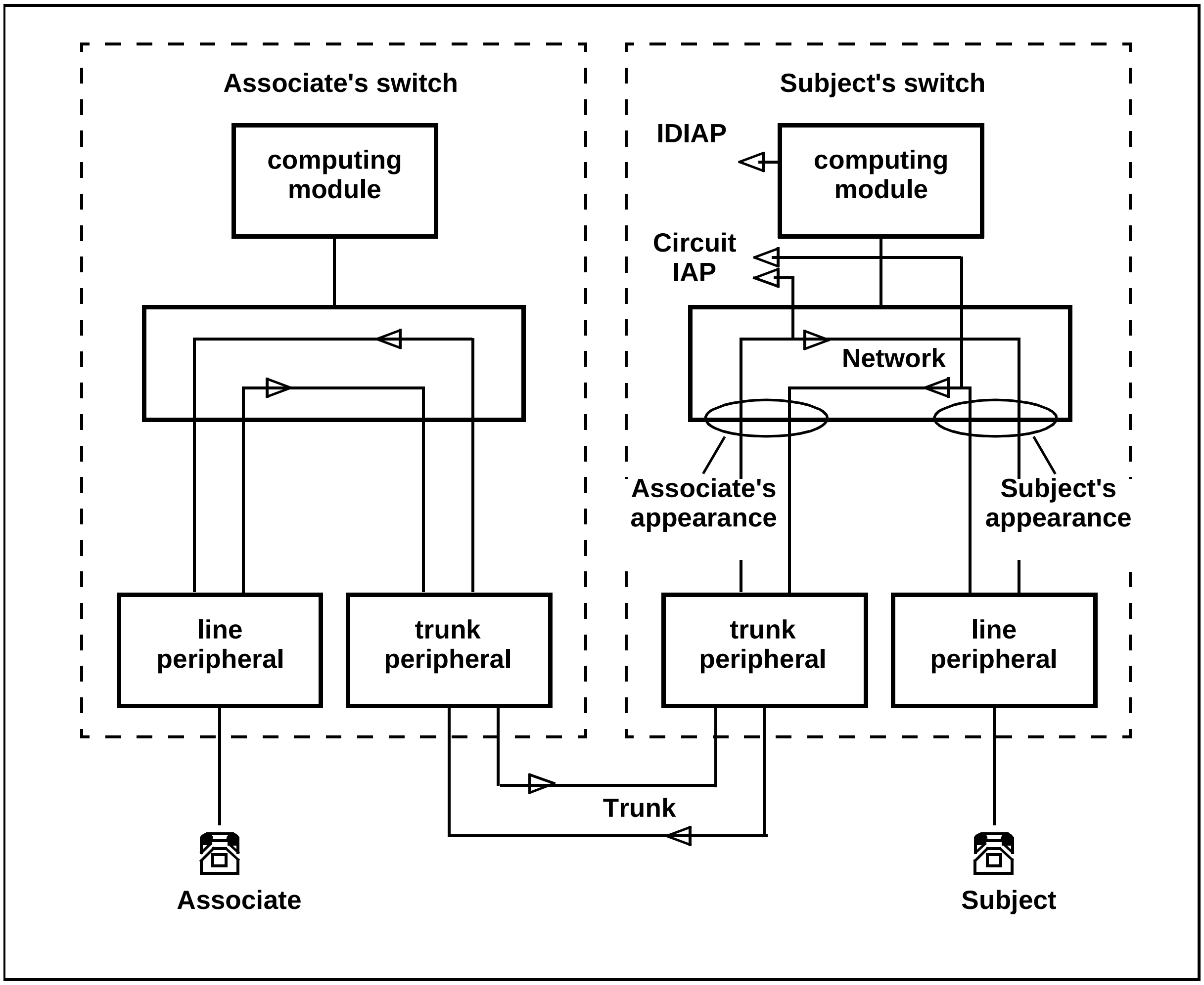
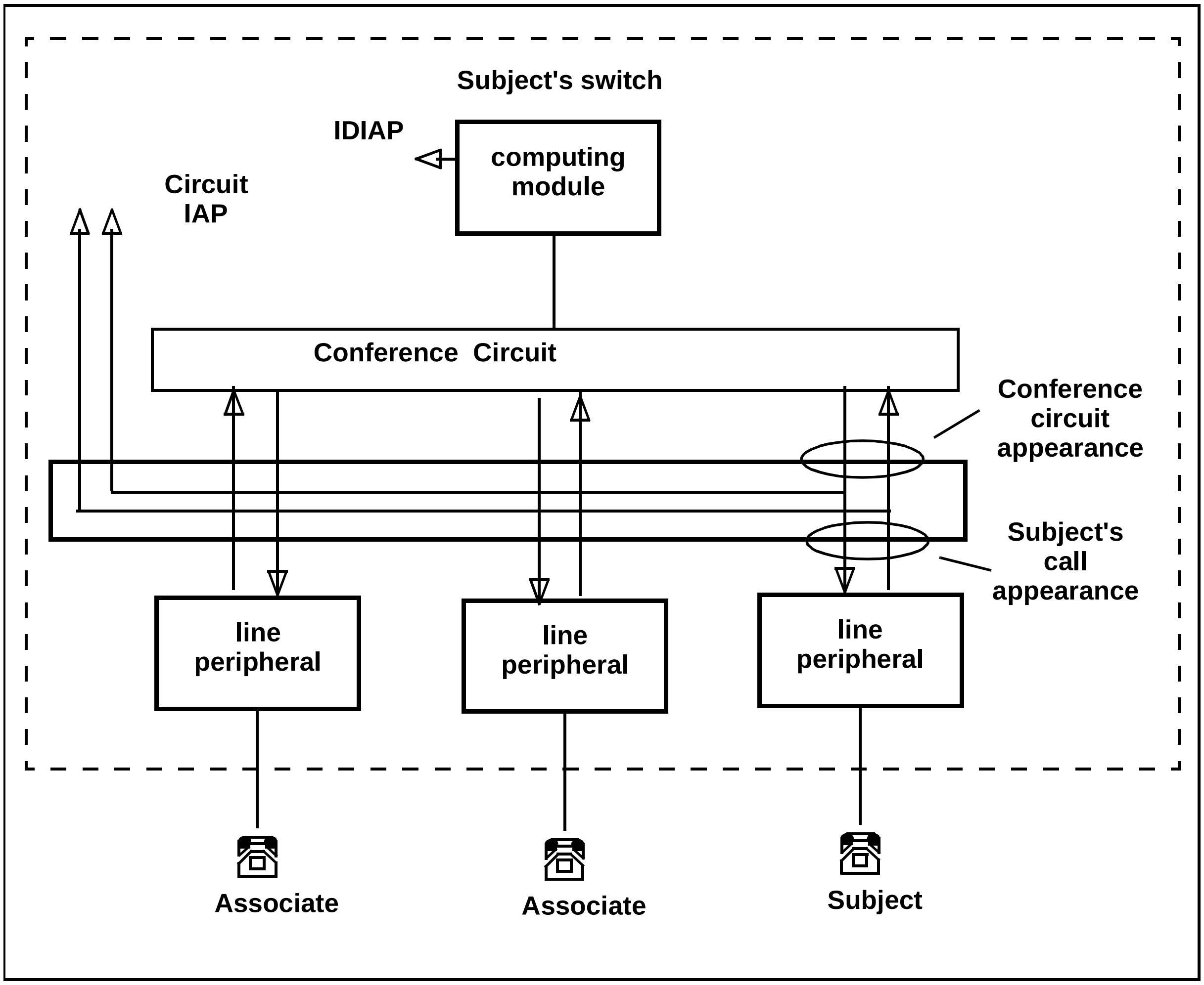


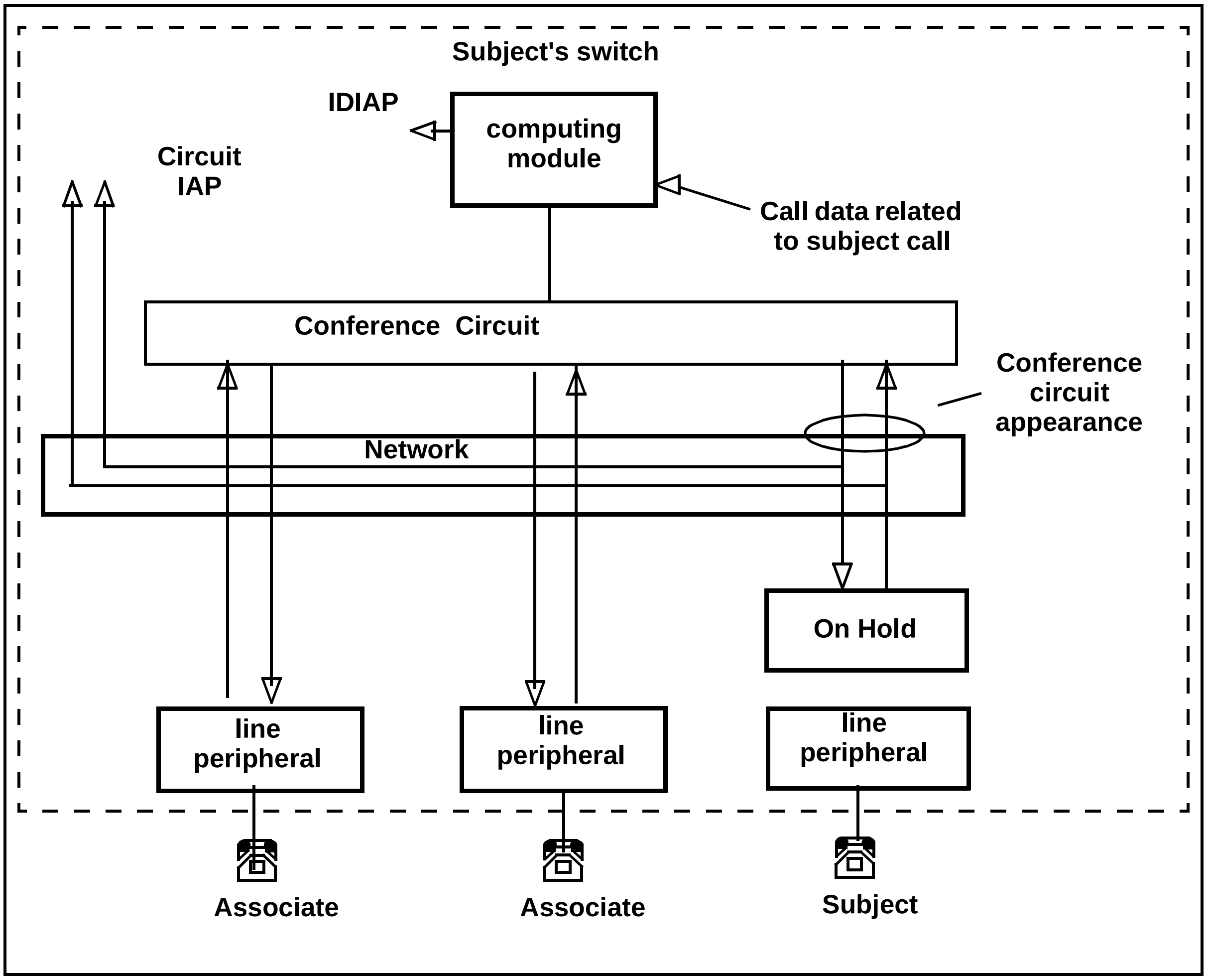

Figure 11-3 depicts a circuit-switched inter-switch call between a subject and associate. At the subject's switch, the Circuit IAP has access to the transmit paths of the subject and associate call appearance at the network. In the case of the associate, the appearance at the network is the outgoing trunk used in the call and not the actual associate's line located on the associate's switch. The Circuit IAP does not have access to any call content available to the associate unless it is present at the subject switch outgoing trunk appearance at the network. For example, call wait tone provided to the associate is not available at the Circuit IAP.

**Figure 11-4: CALEA Three-way Call Access Model**




In the case of a subject-initiated three-way call, the Circuit IAP does not have direct access to call content associated with the associates involved in the conference call but does access the associates' call content through the call leg between the subject's call appearance and conference circuit's appearance present at the network. This is shown in Figure 11-4.

**Figure 11-5: CALEA Three-way Call On Hold Access Model**




### Conference Circuit Intercept Access Point (IAP)

In 502.10 and later generics, the CALEA feature provides the capability for the LEA to monitor the content of conversations by all parties connected through a subject-initiated conference call. This capability is limited to conference call participants that maintain a circuit connection to the call. The content of conference calls is extracted at the Intercept Access Point (IAP). The Connection and Connection Break messages are used to report a change of participants in the subject-initiated call such as situtations where the subject places the call on hold, a party joins the conference, and so on. These messages are reported by the call-identifying information IAP. This is shown in Figure 11-5.

### Packet Data

In situations where packet terminals or applications are deployed, service providers can operate the ISDN system as a DMS-10 exchange termination and an external packet handler system. ISDN D-channel packets are collected within the line drawer of the Line Concentrating Equipment (LCE) and concentrated into a channel that is nailed up through the DMS-10 network. ISDN B-channels that are used for packet service are also nailed up through the DMS-10 network. The nailed-up channels are routed to the external packet handler system.

The CALEA feature supports the interception of ISDN B-channel packet-mode data. Since ISDN B-channel packet data connections (BCON) are connected through the DMS-10 network, they are considered to be circuit-switched and can therefore be accessed by the Circuit IAP.

B-channel packets are treated by the DMS-10 switch as bit streams - the DMS-10 switch does not have access to the data, but transports the packets without any involvement in packet protocol or management. Two CCCs are assigned to this bit stream, delivering both directions of the packet bits to the LEA's collection function. Law enforcement agencies are responsible for interpreting the bit streams and understanding the protocol.

### Nailed-up connections

The CALEA feature supports the interception of semi-permanent (nailed-up) connections. The type of information being passed, either voice or data, is unknown to the DMS-10 switch. Since the nailed-up connections (CONN) are connected through the DMS-10 network, they are also considered to be circuit-switched and can be accessed by the Circuit IAP. When nailed-up connections are intercepted, the information they carry is delivered to the LEA's collection function. Law enforcement agencies are responsible for interpreting the data.

## Call Data Messages

The CALEA feature generates call identifying messages to describe surveillance events during the progress of the call. These messages are formatted using a specialized protocol called Lawfully Authorized Electronic Surveillance Protocol (LAESP). The LAESP messages are transported over a CDC and delivered to an LEA's collection function.

The basic surveillance event messages supported by the DMS-10 switch in 501 and later generics include:

- Answer
- Change
- Origination
- Redirection
- Release

- TerminationAttempt

Messages reporting the assignment and release of a call content channel are also generated and delivered to the LEA. These messages include:

- CCOpen
- CCClose

A ConnectionTest message is used to test CDC connectivity to the LEA. It is generated and delivered to the LEA when the CDC is tested either automatically or manually. The message is also used to inform the LEA that a particular CDC connection was lost and had to be rebuilt. In this event, CDC messages may have been lost. When informing the LEA of a lost connection, the message is sent to the same IP address and port whose connection was lost.

Additional surveillance event messages supported by the DMS-10 switch in 502.10 and later generics include:

- Connection
- Connection Break
- Dialed Digit Extraction
- Network Signal
- Subject Signal

The Connection message reports the addition of one or more participants to an existing call. The ConnectionBreak message reports when one or more participants are removed from an existing call. Together, these messages are used to identify the parties connected to a call at all times.

The DialedDigitExtraction message is generated and delivered to the LEA when digits are dialed by the intercept subject after the call is cut-through to another service provider for processing and routing. Only the subject is monitored for post cut-through DTMF digits. The associate or MRP is not monitored for post cut-through digits because their service is not under surveillance.

The NetworkSignal message reports signals generated or sent by the IAP switch to the intercept subject using the facilities under surveillance. The network signals reported by this message are signals originated and applied by the accessing system IAP toward the intercept subject.

The SubjectSignal message reports subject-initiated signals that control a feature or service operation (such as call forwarding, call waiting, call hold, and 3-way calling). If the user input is uninterpretable and would result in no change in the control of the call, this message may still be generated.

## Call Content Delivery

Call content is replicated and exported through CCCs. No audible indications are provided either to a surveillance subject or to an associate to indicate that accessing and exporting call content is occurring. The CALEA feature does not deny availability of any service to either a subject or associate.

### Call content channels (CCC)

A subject's call content is transported to the LEA through CCCs. The number of CCCs available for an electronic surveillance varies according to the requirements of the LEA. Factors influencing the number chosen include the subject's line type, the subject's call feature capabilities, the type and capacity of individual CCCs, the number of possible call appearances, and the subject's call-related activities. In the DMS-10 CALEA feature, individual CCCs use separate channels for subject transmit and associated transmit paths. Therefore, a pair of CCCs is required for each intercepted call. Each CCC used for an electronic surveillance must be capable of transporting all of the subject's intercepted bearer services.

Additional CCCs, up to the number provisioned for a particular electronic surveillance, are used when the subject uses services allowing multiple call legs, for example, call wait, three-way call, and user transfer.

The DMS-10 switch assigns a bearer capability to a call during call establishment based on the originating interface's characteristics or the subscriber's bearer service. The bearer service cannot be changed after a call has been established. The DMS-10 switch assigns CCCs that can handle a bearer capability greater than or equal to the call's bearer capability. This is controlled by the bearer capability assigned to the Call Content Group (CCG) provisioned for a surveillance.

### Channel delivery methods

The CALEA feature supports only separated call content delivery. The DMS-10 delivers intercepted call content through two independent Call Content Channels (CCCs): one to carry a replication of the transmit path of the Subject, or a monitored replacement party (MRP), and the other to carry a replication of the sum of the transmit contents of all the Associates.

Call content introduced from an associate-controlled call that is not available to the subject or subject's appearance is not accessible and is, therefore, not delivered.

A number of CCCs may be associated with a particular surveillance. This collection of CCCs is referred to as a Call Content Group (CCG). The operating company assigns CCCs to the CCG that can transport all of the subject's intercepted bearer capabilities.

## Dialed Digit Extraction

The Dialed Digit Extraction (DDE) feature for CALEA provides the ability to detect and extract digits generated by a monitored subject for delivery to authorized law enforcement agencies (LEA).

The DDE feature performs the following functions:

- Detects digits generated after call cut-through and sends them to one or more LEAs.
- Administers both Call Content Channels (CCC) and DDE on a per-trunk basis.
- Downloads a DSI pack based on its pack code.
- Enables and disables DDE and CCC trunks.

### Detecting Digits

When digits are dialed by the intercept subject after the call is cut through to another service provider (for processing and routing, for example), this feature generates and delivers a DDE message to one or more LEAs. Only the subject is monitored for post cut-through DTMF digits. Digits are accumulated and reported in the DDE message under any of the following conditions:

- a maximum of 32 digits have been received
- 20 seconds have elapsed since the first digit was received
- the call (or call identity) is released
- a call identity is split into two call identities or two call identities are merged into one call identity (as when a Change message is generated)
- the Meridian Business Set (MBS) Automatic Dial (AUD) feature key is pressed while in the talk state

### CCC and DDE resources

A surveillance is represented by a case identity (CaseID) within the DMS-10. The Case ID is the collection of parts for the surveillance, including the CDC, CCCs, and a list of subject directory numbers (DNs) under surveillance. CCCs are declared within Call Content Groups (CCGs). When call content is not required for a given surveillance, then DDE resources may be declared within a CCG. When call content is required, one of the two CCCs used to provide call content for the call is also used for DDE. After each part of the surveillance is assigned, a Case ID is created to link these parts together into a surveillance.

Digits dialed after cut-through are detected by the NT4T50 CALEA Interface pack. Only DSI digital trunks may be used as DDE resources. When call content is required for a given surveillance, the equipment provisioned for CCCs will also be used to provide the DDE capability if the CCC is linked to an NT4T50 pack.

## DMS-10 Network Interfaces

The Public Switched Telephone Network interfaces carry intercepted voice and data transmissions into the network through digital trunk facilities. The digital trunk facilities provide a full-duplex 1.544 Mbps transmission pipeline. Bandwidth is divided into 8 kbps overhead plus 1.536 Mbps of user information. For digitized voice applications, the information bandwidth consists of 24 multiplexed 64 kbps channels. Each channel can be used to carry voice, data, and circuit-mode data (56 kbps or 64 kbps).

For most CALEA interceptions, the digital trunk interface used may employ the robbed-bit signaling technique, in which bits are borrowed from information channels and used to convey signaling information. Removing a bit from an information channel has no discernible effect on voice quality. Robbed-bit signaling limits the transmission rate information channels to 56 kbps.

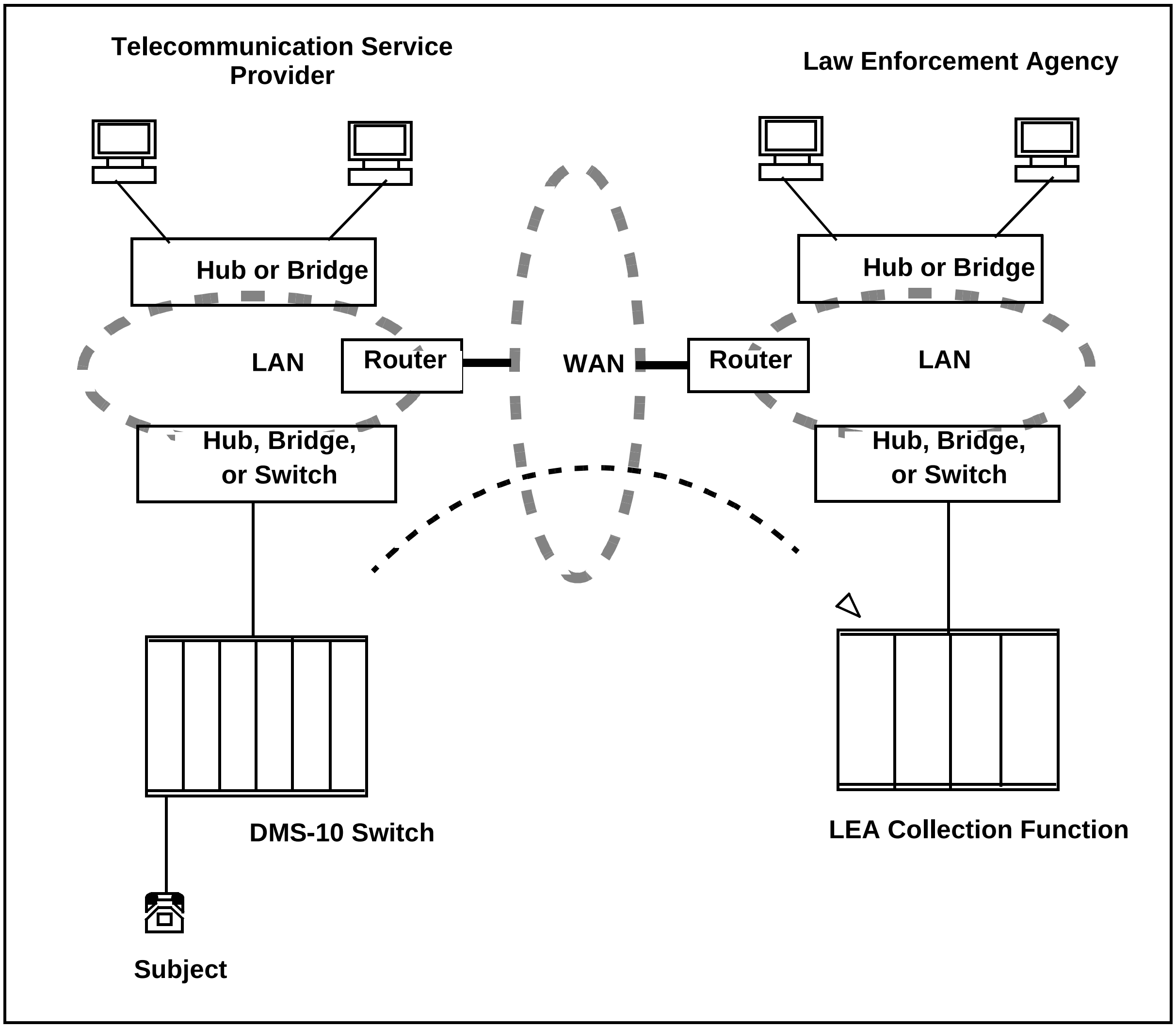
When ISDN circuit-mode or packet-mode data is intercepted at 64 kbps, the digital trunk interface used must be set up to use 64 kbps clear channel capability (using the B8ZS coding scheme for zero code suppression). The 64 kbps clear channel capability must be maintained from the point of origin (the DMS-10 switch), through the network, to the LEA's collection function. The 64 kbps clear channel capability is recommended for interceptions of ISDN circuit-mode or packet-mode data since the data equipment can operate at either 56 kbps or 64 kbps. The DSI module provides call content delivery. Each DSI module contains two digital trunk interfaces that may be configured for either the 56 kbps or 64 kbps transmission rate (64 kbps is required for clear channel capability).

### Wide Area Network (WAN)

A Wide Area Network (WAN) is used for computer-to-computer communications between the DMS-10 switch and LEA monitoring centers. IP routers manage the connections from the DMS-10 switch and the LEA's monitoring centers to the WAN. An Ethernet (RFC894), using the TCP/IP protocol suite, connects the DMS-10 System Processor pack (NT3T98) to a hub, bridge, or switch which is, in turn, connected through a LAN to the IP router.

Figure 11-6 provides an example of CALEA message flow from the DMS-10 switch to the LEA collection function through the WAN. CALEA messages are collected by the DMS-10 switch and then placed in IP packets. The IP packets are sent over the Local Area Network (LAN) to the router. The router then sends the packets over the WAN. At the LEA collection function, a router takes the IP packets off of the WAN and sends them over the LEA LAN. The LEA collection function computer selects packets addressed to it and monitors for any missing packets. Since the packets can arrive in any order, the computer reassembles the packets and converts the data back into the original CALEA message format.

**Figure 11-6: CALEA Message Flow through Data Network**




## Operations, Administration, and Maintenance

The CALEA feature requires that surveillance be controlled by an authorized telecom service provider. Messages controlling surveillances are passed between the authorized telecom service provider (*authorized user*) and the CALEA administration and provisioning function through the Service Provider Administration Function (SPAF). SPAF is performed either through a standard DMS-10 teletype (TTY) using an RS-232 port, or through a virtual TTY using the Ethernet port on the NT3T98 System Processor pack. One or more TTY ports can be dedicated to SPAF. The TTY ports cannot be use for shared maintenance, administration, or CALEA functions. The TTY ports may be accessed locally or remotely using modems.

The authorized user sends messages to the subject switch through SPAF, to control the setup, activation, modification, and deactivation of surveillance. Surveillance-related output messages (messages sent from the DMS-10 switch to the surveillance administration) are only output to TTYs assigned a Lawfully Authorized Electronic Surveillance (LAES) class password.

### CALEA administration

Administration tasks including setting up the CALEA feature, and establishing and controlling surveillance functions are performed through Overlay SURV. The SURV overlay can be accessed only through two password-protected CALEA access levels, surveillance administrator level and surveillance user level. There can be only one administrator per switch but more than one user. Administrator-level tasks must be initiated through surveillance administrator-level access; surveillance provisioning and management tasks must be initiated through surveillance user-level access.

The administrator-level tasks that can be performed, using Overlay SURV, include:

- adding and deleting surveillance user-level identifiers (USIDs)
- displaying a list of surveillance user IDs
- resetting a password associated with a user ID
- changing the surveillance administrator's password
- setting an abort timer for SURV overlay inactivity

There is a maximum of 20 assignable surveillance users or user IDs. User-level tasks that can be performed include:

- changing their passwords
- assigning call content channels (CCC), call data channels (CDC), and call content groups (CCG)
- assigning Case IDs
- activating/deactivating surveillances
- testing call data channels
- verifying digital trunk provisioning for DDE
- assigning digital trunks to be used for DDE

### CALEA maintenance

Standard DMS-10 maintenance and diagnostic procedures are used for DMS-10 systems or circuits used by the CALEA feature.

**CCC maintenance** When maintenance activities or a system fault cause a subject or associate station to be placed in a different condition, either spared or non-spared, a call on the station is re-connected, if possible, using a different path through the network. The DMS-10 switch also attempts to re-connect the CCC channels being used for the subject.

When a peripheral loop controlling a CCC connection is taken out of service, the CCC connection is dropped. The subject and associate are not dropped or re-switched. When one CCC of a pair is taken out of service, the monitoring continues on the remaining half of the conversation. When a diloop is taken out of service, an attempt to move the CCC connection to another diloop is made. The subject and associate are not re-switched unless their two-way conversation is occurring over the defective loop. Any form of sparing/reswitching may cause audible clicks on the line.

**CDC maintenance** The DMS-10 switch provides an audit function for both connected and unconnected CDCs to determine whether the IP addresses and port numbers declared in Overlay SURV agree with the IP addresses and port numbers used to establish the connection. An audit function also verifies that no connections are maintained for inactive CDCs.

When problems such as network failures or LEA collection resets cause CDC connections to be lost, the DMS-10 switch attempts to re-establish the connection.

**Alarms** Alarm teletype messages are generated in the event of CALEA-related failure such as a lost CDC connection, at both the maintenance-class TTY and at the LAES TTY. CALEA-related failure messages are output at the LAES TTY only when the TTY is logged on. The teletype messages do not identify the subject being monitored. Alarm messages are also generated for problems with CCC trunks, but display only at the maintenance-class TTY. Alarm messages, once output, are not stored internally for later retrieval.

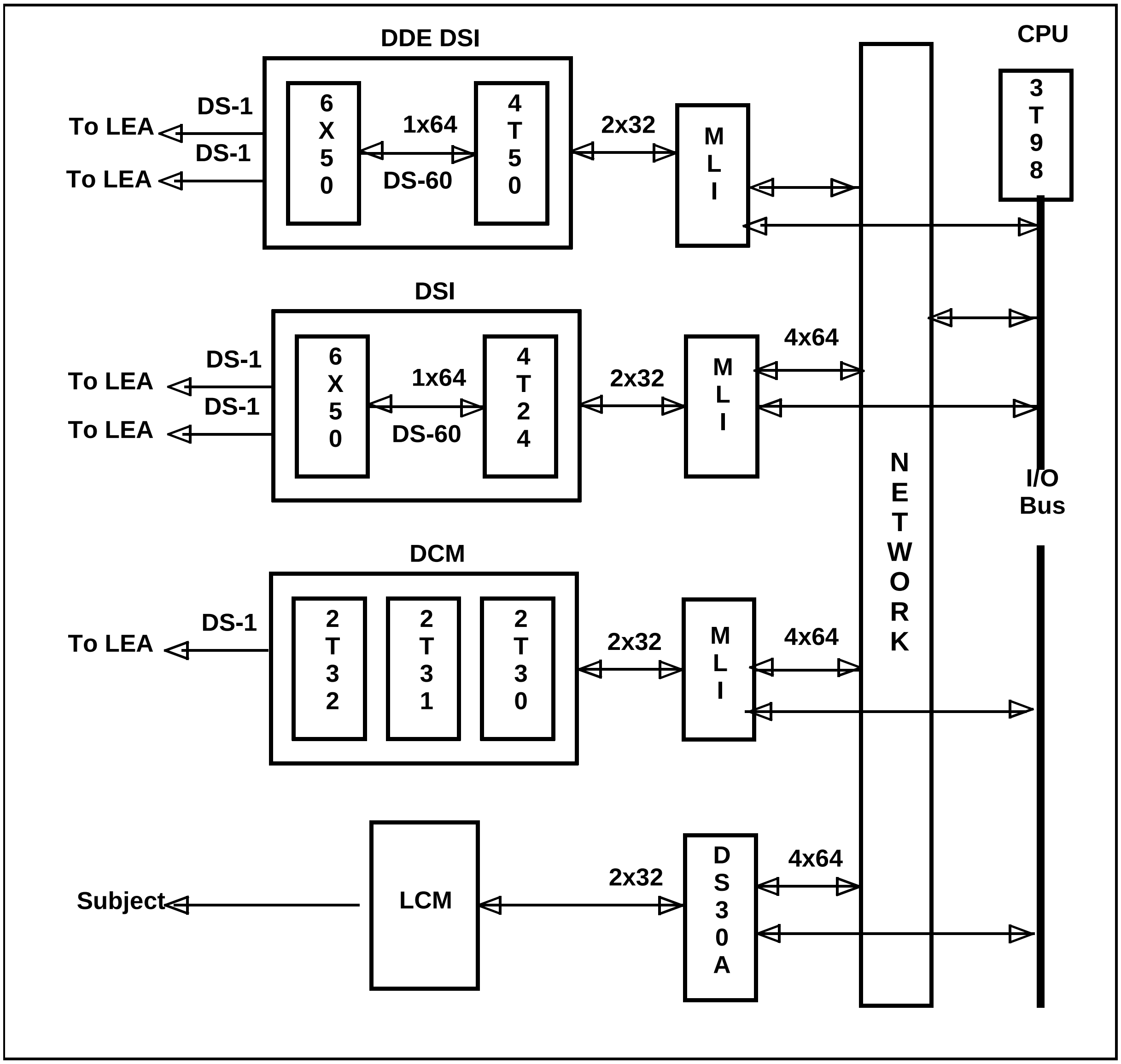
## DMS-10 CALEA System Architecture

Figure 11-7 shows the DMS-10 CCC architecture that supports the CALEA feature. The main components include:

- System Processor pack (NT3T98)
- Network pack (NT4T06) (DMS-10 Classic Network configuration)
- DS-30A Interface pack (NT4T04) (DMS-10 Classic Network configuration)
- Multiplex Loop Interface (MLI) pack (NT4T05) (DMS-10 Classic Network configuration)
- Network pack (NT8T06) (DMS-10EN network configuration)
- Network Interface pack (NT8T04) (DMS-10EN network configuration)
- Peripheral trunk equipment

- Digital Signal Interface (DSI): NT4T24 and NT6X50
- Digital Carrier Module (DCM): (NT2T30, NT2T31, and NT2T32)
- CALEA Interface: (NT4T50)

**Figure 11-7: DMS-10 CALEA System Architecture**



## CCC Message Flow

There are four basic intercepted call types:

- intra-office (line-to-line) circuit-switched voice/data
- inter-office (line-to-trunk and trunk-to-line) circuit-switched voice/data
- nailed up connection voice/data
- ISDN B-channel packet-mode data

When transmit paths (subject's path and associate's path) for the four calls types are intercepted, they are switched to the DSI or DCM module responsible for delivering the intercepted call content on separated CCCs. Once the circuit has been established by call control signaling, no processing is required within the DMS-10 switch until the call is cleared or affected by the subject's services.

For each call type, the basic call components that comprise the DMS-10 CALEA architecture include:

- subject's and associate's customer premises equipment (CPE)
- Digital Signal Interface (DSI): NT4T24 or NT4T50, and NT6X50
- Digital Carrier Module (DCM): (NT2T30, NT2T31, and NT2T32)
- Line Concentrating Module (LCM)
- System Processor pack (NT3T98)
- circuit-switched network
- packet handler and packet-switched network

These components are connected through time-division multiplexed (TDM) loops. The TDM loops each contain individual channels of the following types:

- DS0 - one information channel transmitted at the nominal rate of 64 kbps (equivalent to one voice circuit)
- DMSX - signaling channel between the MLI pack (NT4T05) or Network Interface pack (NT8T04) and the peripheral equipment (DSI/DCM)

The loop types include:

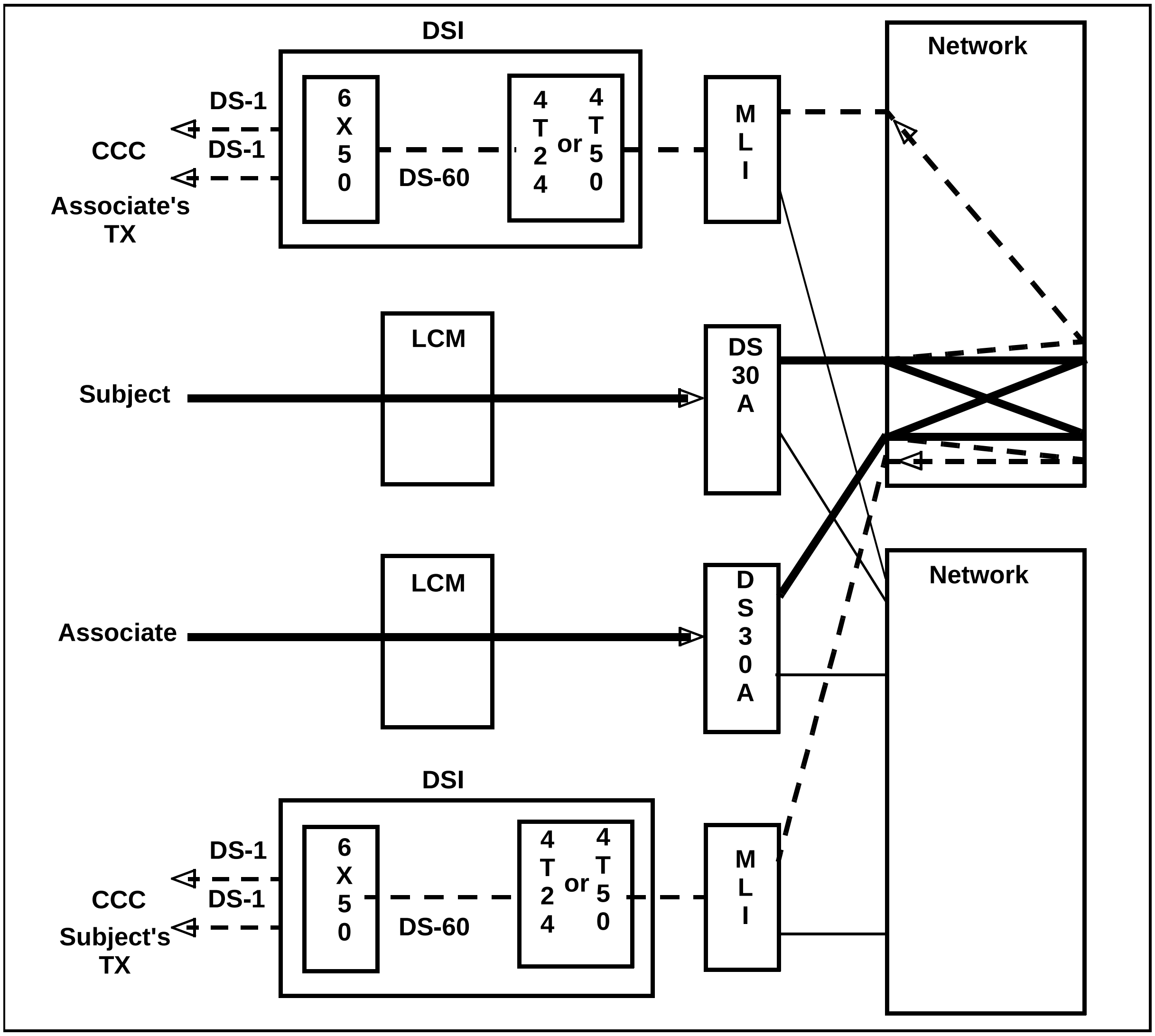
- DS1 - contains 24 * 64 kbps channels and 8 kbps for framing bits, giving a total rate 1.544 Mbps, used by the North American T1 carrier
- DS-30A - contains 32 * 64 kbps channels, with a total rate of 2.56 Mbps. The first channel on the primary loop is allocated as the DMSX communications channel

- MLI - contains 32 channels, with 24 channels mapped to the 24 channels on the DS1 loop. Any number of these channels can be semi-permanent (nailed-up) connections; the remaining channels are unused. The rate of the MLI loop to the DSI is 64 kbps; the rate of the MLI loop to the DCM is 56 kbps.
- DS-60 - contains 64 * 64 kbps channels, and is used between the NT6X50 and NT4T24/NT4T50 packs of the DSI. Forty-eight channels from the two DS1 loops are mapped to the 48 channels on the DS-60 loop; the remaining channels are unused.

## Communication Interception

Figure 11-8 illustrates call content extraction within the network of the DMS-10 switch, by the time-space switch matrix. Surveillance, as shown in this example, is performed using one-way reproduction of the transmit path through the MLI and DSI, and out to the trunk used for a Call Content Channel (CCC).

**Figure 11-8: DMS-10 CALEA Communication Interception**



## Dialed Digit Extraction

Dialed digit extraction (DDE) is performed by a digital signal processor (DSP) located within an NT4T50 DSI module. Like call content extraction, DDE requires a one-way reproduction of the subject's transmit path. The associate's transmit path is not required unless the surveillance is set up for call content delivery.

## Provisioning Guidelines

Recommendations for provisioning traffic-related DMS-10 equipment are found in NTP 297-3601-450, *Provisioning*.

### Call Content Channels (CCC)

Provisioning CCCs entails ensuring that adequate network and digital trunk equipment have been installed to meet the operating company's and LEA's requirements for a surveillance interval. Thus, CCCs should be included in any digital peripheral shelf calculations and in the calculations for the number of peripheral loops, network interface packs, network packs, and network shelves.

In order to ensure network availability at a constant grade of service when CCCs are required by different LEAs, it is recommended that peripheral equipment used by each LEA is assigned to different network interface packs. In addition, CCCs should always be assigned in even numbers since two CCCs are required for each intercepted call.

### Dialed Digit Extraction

Only NT4T50 DSI trunks can be used as DDE resources. DDE is required for all surveilled calls, intra-office as well as inter-office. When call content is required, the DSI equipment provisioned for CCCs is also used to provide the DDE capability.

CCGs may be dedicated to a specific surveillance or shared by a number of different surveillances. When call content is not required, then DDE resources may be declared within a CCG and the CCG will be used for DDE purposes only. It is recommended that one CCG be used for DDE purposes and that it should be shared by all surveillances (for a given LEA) which do not require call content.

Sufficient DDE resources must be assigned to satisfy all active surveillances. In the event that a DDE resource is needed and none is available for a surveilled call (group resource exhaustion), an indication will be provided to law enforcement in the release cause parameter of the release message. See NTP 297-3601-450 for additional information on provisioning DDE resources.

### Call-identifying information data channel (CDC) components

The main components of the CDC include:

- System Processor pack (NT3T98)
- Ethernet hubs, bridges, or switches
- routers

Call identifying information is identified by the NT3T98 software applications, formatted into CALEA messages, and is transported from the DMS-10 switch to the LEA's collection function using the Ethernet interface on the NT3T98 pack. Since duplicate NT3T98s are provisioned in a DMS-10 switch, should the active NT3T98 fail, the standby NT3T98 automatically assumes control.

Hubs can be used to create a cluster of nodes, such as a DMS-10 site, that are then connected to the LAN. This capability enables DMS-10 sites to then be added to the LAN as the network grows. Two hubs can be provisioned, for redundancy. Each hub should support multiple RJ-45 modular ports for supporting 10/100BASE-T host connections over unshielded twisted pair cabling.

Two routers should be configured, for redundancy. Since some routers support a number of Ethernet ports, the need for hubs may not exist. The routers, in that case, connect directly to the NT3T98 pack.

### DMS-10 network options

The CALEA feature is supported both by the DMS-10 Classic Network and by the DMS-10EN Network. The DMS-10 Classic Network has a call blocking structure. As the number of active calls increases relative to the overall network capacity, the more likely call blockage is to occur. Each active surveillance requires additional connections in the network that may increase the possibility of call blockage. Switches with DMS-10 Classic Networks that encounter these blocking conditions may need to be upgraded with a larger network configuration or with the DMS-10EN Network.

# DMS-10 CALEA System Limitations

## Call Content Loss

Call content is not delivered to the LEA's collection function under the following conditions:

- network paths to CCCs could not be established within the DMS-10 Classic Network
- facilities or equipment used for the CCCs are either not provisioned correctly or are out of service
- facilities or equipment used for the CCCs are not capable of transporting one or more of the subject's intercepted bearer services
- all CCCs provisioned are in use
- switch capacity for the total number of simultaneous intercepted calls is exceeded
- switch capacity for the number of CCCs per final call leg is exceeded. This may occur, for example, when a call is forwarded and each party (calling party, all forwarding parties, and the final called party) is under surveillance.

- the subject is located off of a remote equipment module that is in Emergency Stand Alone (ESA) mode
- maintenance activity performed on equipment used for call content prevents delivery
- switch resource exhaustion. This may occur, for example, when a call involving the subject contains the last call register, leaving no call register available for the CALEA feature to monitor the call.

Between call completion (answer) and call release, call content may still be lost (that is, the call content is truncated). Call content is not delivered under the following conditions:

- network paths to CCCs could not be established within the DMS-10 Classic Network
- switch capacity for the number of surveillances on a single call leg is exceeded. For example, if a call is transferred by the subject and the subject then drops out of the call, the two call legs are combined on one call leg; the combined surveillances exceeds the switch capacity.
- the subject is located off of a remote equipment module that enters ESA mode
- maintenance activity performed on equipment used for call content requires the connection to be dropped
- a site restart is performed while call content is being delivered to the LEA's collection function

Call content will not be provided for audible tones or recorded announcements that are used to convey call progress or call failure information to the subscriber. Tones provided between call completion and call release by the DMS-10 network equipment, such as call waiting tones, interrupt the two-way speech path between calling parties and, thus, are not provided in call content.

## Call-identifying Information Loss

Call-identifying information is not delivered to an LEA's collection function under the following conditions:

- facilities or equipment used by the CDCs are either not provisioned correctly or are out of service
- switch capacity for the number of simultaneous calls is exceeded
- the subject is located off of a remote equipment module that enters ESA mode
- a site restart is performed while a call is being monitored

- CDC connections could not be established due to the LEA collection function not accepting new connections
- switch resources are all in use
- a CDC is congested and the associated buffers are full

The call-identifying information for DDE is not delivered to an LEA's collection function when:

- all digital trunks provisioned for CCC or DDE are in use
- a network path to a digital trunk provisioned as a DDE resource could not be established within the DMS-10 Classic network
- maintenance activity performed on equipment used for DDE prevents digit reporting
- the CCG is provisioned with DCM or DSI (NT4T24/NT6X50) equipment, which lacks the DDE functionality. DSI equipment (NT4T50/NT6X50) must be installed.
- a Meridian Business Set (MBS) subject calls another MBS user when both parties are located on the same DMS-10 switch. In this case, no DTMF tones are sent to the MBS user.

## TEEN Secondary DNs

The TEEN feature allows up to five directory numbers to be assigned to a single-party line on a station - a primary DN (PDN) and four secondary DNs (SDN). Although calls can terminate to either the PDN or to the SDN, they can originate only from the PDN. Thus, surveillance can be activated only on the PDN; the surveillance is inherited by the SDNs.

## ESA Mode in DMS-10 Remotes

CALEA surveillance is not supported on remotes that are in Emergency Stand-Alone (ESA) mode, because all communication with the host, where the IAPs are located, is broken.

## Large Cluster Controller (LCC)

The DMS-10 LCC is a host office that supports a greater cluster line capacity than a Host Switching Office (HSO). Since the LCC does not perform telephone switching functions, it cannot support CALEA functions.

## Area Code Boundary

In configurations that support 7-digit dialing across an area code boundary, it is possible for the wrong NPA to be appended to the called party's number. For example, when the subject dials a number that exists in an office located in a different area code and is not required to dial the NPA, the DMS-10 switch appends the caller's NPA, rather than the called party's NPA, to the called party's number. Thus, the Called PartyIdentifier parameter sent to the LEA will contain the subject's NPA and called party's DN.

The LEA should determine, therefore, whether 7-digit dialing across an area code boundary is supported before activating surveillance on a subject. When 7-digit dialing across an area code boundary is supported, the operating company must set the terminating party's NPA through translations by using the *Set NPA* node (SNPA) in the appropriate translation screen (see Overlay TRNS in NTP 297-3601-311, *Data Modification Manual*).

## Inter-office Signaling and Party Identity Parameters

When an associate's appearance is a trunk from another office, the calling party's DN is identified in the Calling PartyIdentity parameter when it is known by the DMS-10 switch; otherwise, the TrunkID parameter identifies the incoming (or two-way) trunk group used.

The calling party's DN may be provided to the DMS-10 switch when ISUP signaling and/or SIP packet trunks are used from the originating switch through the Public Switched Telephone Network to the DMS-10 switch. If an in-band signaling trunk is used at any point in setting up the call, then information about the calling party DN is lost.

When the associate's appearance is a trunk to another office, then the answering party's identity is not known.

When the associate's appearance is a trunk from another office, the original called party and last redirecting party identities are included in the RedirectedFromInformation parameter, when known by the DMS-10 switch.

## I-Megabit Modem Service (1MMS)

1MMS provides a high-speed data connection used to carry TCP/IP Ethernet messages between the customer premises 10Base-T network and an operating company-based 10Base-T network hub, router, or packet switch. Since the data content bypasses the DMS-10 network, it cannot be placed under surveillance.

# Section 12: Index

## Numbers-

# D

# F

## L



## M

# N

# T

DMS-10 Family

# 600-Series Generics

Feature and Services Description



NTP number: NTP 297-3601-105
Release: 08.01
For Generic 602.20
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NORTEL