



TECHNICAL REFERENCE

Special Access Connections

To The

AT&T Network

OCTOBER 1996

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Special Access Connections

To The

AT&T Network

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**Special Access Connections
To The
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**Part I
General Information**

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1. SCOPE

This document provides a description of the various special access interfaces at the AT&T point of termination (POT) for AT&T services. This document updates and replaces in its entirety the previous issue of AT&T Technical Reference 41458^[1].

This is not a Customer Premises Equipment (CPE) requirements document, although some of the material may affect CPE design objectives. Since the boundaries of AT&T services are the AT&T points of termination, this technical reference applies only to the signals crossing the interfaces at those points. In addition to the interface specifications, some CPE guidelines are included as an aid for CPE vendors.

This reference describes the special access connections of CPE to an AT&T Action Point (ACP). Switched access (i.e., access which is routed through a Local Exchange Carrier's central office switch) is not covered in detail, but is mentioned. Interface specifications for private switched services, such as Enhanced Private Switched Communication Systems (EPSCS) and Common Control Switching Arrangements (CCSA) are not covered in this document, but are instead covered in AT&T Technical Reference 43252^[2]. The Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) interface specifications are provided in AT&T Technical Reference TR 41459^[3], and are not covered here. However, TR 41459 refers to this document for information concerning transmission levels.

2. CHANGES SINCE LAST ISSUE

Additions to this issue of TR 41458 include:

- AT&T Toll Free Multimedia Service,
- Universal T1.5 Access,
- ANI Delivery for SDN, MultiQuest[®] and Toll Free MEGACOM[®] users,
- Elimination of AT&T World Connect Service,
- Enhanced descriptions of AT&T's Switched Digital Services.

3. ORGANIZATION OF THIS DOCUMENT

Part I contains general and introductory matter, as well as a glossary of terms and acronyms. Also included in Part I are descriptions of various access services, features and dialing plans.

Part II contains information regarding access interface to an Action Point (ACP). Only the behavior of the interface at the point of termination is described.

Part III contains recommendations for CPE designers, based on certain assumptions about probable access arrangements (LEC, CPA, etc.) which connect CPE to an ACP. If these assumptions are not appropriate (e.g., if customer-provided access arrangements are used), some sections of Part III may not be applicable. More detailed information on access arrangements is contained in AT&T Technical Reference TR 622^[4].

4. SERVICES

AT&T switched services are those services which are offered under FCC Tariffs 1, 2, 4, 9, 11 and 12. The AT&T switched nodal services within this document's scope are Software Defined Network (SDN), Software Defined Data Network (SDDN), Software Defined Data Network International (SDDN-I), MEGACOM[®], Toll Free MEGACOM[®], Switched Digital Services (SDS), Switched Digital International (SDI), Long Distance Service (LDS), Alternate Egress Arrangements, Toll Free Multimedia Service, Universal T1.5 Access and MultiQuest[®].

Table I - 1 lists the access interfaces available for the switched services.

Special access facilities, which extend from the customer's premises to the AT&T service node, must be obtained independently of the nodal service. Special access can be obtained through one of the following arrangements:

1. The customer may request AT&T Total Service. Under this arrangement, AT&T will arrange for the access facilities for service to the customer premises, which falls under the auspices of Tariff 11. Advantages to the customer include (a) AT&T-engineered access, (b) end-to-end service support from AT&T, (c) a single point of contact for maintenance, and (d) a single bill.
2. The customer may request AT&T Access Coordinated Service. Under this arrangement, AT&T performs all of the functions as with AT&T Total Service except that the customer receives the bill for the special access facilities directly from the Local Exchange Company.

Brief descriptions of the services, their features, and dialing plans relevant to them are presented below. It is stressed that dialing plans presented in this document apply at the network interface; the numbers dialed by a user from a station behind the terminating CPE, i.e., the first customer switching vehicle between AT&T and the customer, may differ (according to CPE-supported routing alternatives), and are outside the scope of TR 41458. In all cases, the following dialing plan notation is used:

X Any digit having value 0 through 9.

N Any digit having value 2 through 9.

0/1 A digit having value of either 0 or 1.

NPA (Numbering Plan Area). Often, NPA refers to a geographical division defined by the familiar term "area code". NPAs are currently in the form of "NXX". A few NPA codes within the North American Numbering Plan have been assigned for special uses (such as 8YY for "Toll Free" service, where "YY" is a multiple of eleven) and are known as Service Access Codes (SACs). The 700 SAC has been set aside for any inter-exchange carrier to use for any purpose such as identifying special access endpoints

CC The 1, 2, or 3-digit country code used in international dialing.

NN The national number, can be a four to fourteen digit national number used in international dialing, where the combination of CC+NN is a 7-15 digit number.

1. For example, SDS 56 special access endpoints are assigned numbers with the "700" SAC.

Table I - 1. Availability by Service of Special Access Interfaces.

INTERFACE AVAILABILITY							
SERVICE	2WLRB	2WLS	2WGS	4WE&M	56Kbps	DS-1**	ISDN PRI
SDN		X*	X*	X	X	X	X
SDDN					X	X	X
MEGACOM		X	X	X		X	X
Toll Free MEGACOM	X	X	X	X		X	X
Switched Digital Services					X	X	X
Switched Digital International					X	X	X
Alternate Egress Arrangement						X	X
Toll Free Multimedia Services							X
MultiQuest	X	X	X	X		X	X
* 2W Interfaces are only supported for the Express Connect feature							
** Robbed-Bit Signaling							

4.1 Software Defined Network (SDN) and Software Defined Data Network (SDDN)

Software Defined Network (SDN) provides many unique features and a service alternative to private network offerings. SDN makes use of the AT&T Switched Network (ASN) to offer premises-to-premises voice and data transport, along with a rich set of customer controllable call management and monitoring features.

Using SDN access arrangements, users may place on-net calls, off-net calls, and calls to and from private networks. Three types of special access are available between a customer location and the AT&T Network: direct analog access with E&M supervision, T1.5 access using robbed-bit signaling or ISDN PRI. These are

specified in Part II of this Publication. Switched access to SDN is available through a Class 5 office or Centrex.

Software Defined Data Network (SDDN) is a circuit switched data feature of SDN. SDDN provides synchronous full duplex channels at 56, 64, 384 and 1536 Kbps. In addition, nx56, nx64 and nx384 Kbps connections may be dialed using inverse multiplexers on the customer's premises.

Several access options (Section 0) and off-net dialing options (Section 4.1.4) are available. SDDN uses the same switching and transport infrastructure as Switched Digital Services (SDS, Section 4.4). SDDN-I (International) uses the same network capabilities as Switched Digital International (SDI, Section 4.5).

4.1.1 Special Features

Below is a list of some of the call management features currently available to SDN customers (voice only features are marked with an asterisk *). Detailed descriptions of these and additional features available may be found in the AT&T Business Communications Services Guide

- **Originating Call Screening.** Calls originating from SDN stations may be screened for allowable destinations (such as on-net stations, off-net stations, etc.). This screening may vary by originating station group, time of day, or day of week.
- **DID Capability.** SDN will provide Direct Inward Dialing (DID) capability for PBXs that are equipped to handle DID received via special egress connections, but do not have DID for POTS traffic from the local office.
- **Caller Dialed Authorization Codes*.** Caller dialed authorization codes entered via a TOUCH-TONE pad may be used to override restrictions placed on the originating SDN station. An authorization code allows the call to receive a call treatment different from the treatment associated with the originating station. In addition, caller dialed authorization codes may be used by customers for internal accounting purposes.
- **Announcements.** Standard network tones and announcements are part of the basic service. Special SDN-specific recorded announcements can be played to notify the caller of special conditions (e.g., blocking due to screening, request for authorization code, or request for additional digits.)
- **Answer Supervision.** Answer supervision indicates that the called party has gone off hook. This capability may be passed back to those customer PBXs desiring it.
- **Voiceband Data*.** SDN will support voice and voiceband data communication. High performance standards with voiceband data can only be assured when direct access and direct egress are used, and only up to the first point of customer provided switching.
- **Interconnect Arrangement Between SDN and Private Networks.** This interconnect arrangement will provide a dedicated connection between SDN and a private network switch .
- **NXX Sharing Between Customer Locations.** SDN NXX sharing allows an SDN customer with a 7-digit dialing plan to use the same NXX prefix for different physical locations .
- **Location-Dependent Blocking.** Location dependent blocking allows an SDN customer to define a list of numbers that may not be called from a given SDN location. It is not a time-dependent feature. These numbers can be on-net, off-net, or IDDD numbers and are associated with the originating location groups.

Part I - General Information

- Network Remote Access*. Network remote access provides toll-free access to SDN stations from non-SDN stations. This includes public telephones as well as remote customer locations without enough traffic to justify dedicated access lines to the SDN.
- Off-net Overflow Privilege*. The off-net overflow privilege allows automatic overflow from all dedicated SDN direct egress lines or dedicated SDN switched egress trunks to off-net egress routing.
- Forced On-Net. Allows customers to dial a PSTN number to reach a private network destination.
- Authorization code by call type*. If this feature is selected, the Network can prompt a caller for an authorization code when the requested call type is off-net, international, or international off-net.
- Express Connect. Express Connect is the Automatic Connection Service feature which provides a SDN customer the ability to connect two-pre-designated SDN stations together automatically with a simple circuit service. The feature is available to both basic and custom SDN customers. The customer requires dedicated access/egress. By using non-dialable destination numbers in the SDN customer record, unwanted calls (such as wrong numbers) are prevented from reaching the end-user's locations.
- ANI Delivery (see Section 4.3.1)

4.1.2 Dialing Plans

SDN customers may choose a dialing plan comprised of 7-digit numbers, PSTN numbers (i.e., 10-digit numbers for WZ1 destinations or CC-NN format numbers for international destinations), or a combination of the two. Regardless of which dialing plan is chosen, calls may be placed both to on-net and off-net destinations. Salient SDN dialing plan options are listed below.

- A special access SDN customer can place voice calls to on-net and off-net destinations using the dialing plan shown in Table I - 2.

Table I - 2. SDN Special Access Dialing Options.

Dial Plan	Destination (Egress)			
	Domestic Special Egress	Domestic Switched Egress	International WZ1 Egress	International (NWZ1) Egress
7-Digit Private No.	On-net	VON	On-net VON	On-net VON
10-Digit (NANP) 1+10D (1+NANP)	Forced On-net	Off-net	Off-net Forced On-net	-NA-
011+CC-NN	-NA-	-NA-	-NA-	Off-net Forced On-net

- CPE&A for special access/egress to SDN may append a Station Group Designator (SGD) to the called station address digits. This suffix is a one-digit code, whose value may be from 0 to 7, by which the CPE&A may request special screening, special features, or call routing over special Network facilities. SGD-flagged stations will receive the same special feature treatment on every call originating from that station over the same facility. An SGD digit will also be transmitted

from the Network Action Point to the CPE&A on those trunks classified as SGD-capable. The SGD feature is only available on 4-wire E&M and DS-1 signaling trunks from a senderized CPE&A using DP or DTMF address signaling. The Network ACP may require that a CPE&A transmit the SGD code on only a particular (fixed) subset of the trunks. On these trunks, the CPE&A must transmit the extra digit on every outgoing (i.e., from the CPE&A) call.

- Dialing plan for SDDN Special Access.
 - ◇ For static access provisioning, callers may dial on-net and off-net data locations (SDDN, SDS, or international) using the options shown in Table I - 3.
 - ◇ For dynamic access provisioning, callers may initiate end-to-end 56Kbps digital transmission on a per-call basis using a caller-dialed feature code of 115 or by pre-defining a group of stations behind the digital PBX and sending a SGD digit to the SDN network (using CPE&A for special access to SDN, discussed above). With the feature code, the dialing plan is of 115+7D format, where 7D is a 7-digit private number; in the case of the 115 feature code, 10-digit and CC-NN format numbers cannot be used.

Table I - 3. Special Access Dialing Options for SDDN.

Dial Plan	Destination (Egress)			
	Domestic Special Egress ¹	Domestic Switched Egress	International WZ1 Egress	International (NWZ1) Egress
7-Digit Private No.	On-net VON ²	VON ³	On-net VON	On-net VON
10-Digit (NANP) 1+10D (1+NANP)	Forced On-net VON ²	Off-net ³	On-net ⁴ Off-net ⁴ Forced On-net ⁴	-NA-
700-56X-XXXX 700-73X-XXXX ⁵	Forced On-net Off-net	Forced On-net Off-net	-NA-	-NA-
011+CC-NN ⁴	-NA-	-NA-	-NA-	Off-net Forced On-net (future)

Notes:

Figure I - 3 shows the five SDDN off-net egress options currently supported. In these cases, the caller is an SDDN customer and the called party is off-net.

1. SDDN customers participate in multi-point video conferences using AT&T's Global Business Video Services by dialing a 7 or 10-digit VON or by using SDDN 700 Out-Dialing; the video bridge is treated as a Domestic Special Egress destination.
2. The Virtual On-Net (VON) capability allows SDDN customers to assign a private number (7 or 10 digits) to an SDS customer or another SDDN customer.
3. Digital Switched Egress allows the SDDN customer to dial either the NANP (NPA-NXX-XXXX) number or a VON. Digital Switched Egress destinations may or may not be provisioned for SDDN switched access.

Part I - General Information

4. SDDN-I is the international off-net egress option. The SDDN caller dials an International PSTN number (a 10-digit NANP number for IWZ1 or a CC-NN format number for non-WZ1) or a 7-digit Private Number, and the call is routed off-net via Switched Digital International (SDI) service (Section 4.5) and completed using in-country capabilities. When introduced, the Global Software Defined Data Network (GSDDN) capability will allow SDDN callers to place international on-net-terminating calls.
 5. SDDN 700 Out-Dialing allows SDDN customers to dial SDS customers or other SDDN customer locations that have 700 numbers.
- The SDN customers can obtain switched access (via a LEC EAEO) to their SDN network for voice and/or data traffic. The SDN Switched Access/Egress is available only where Feature Group D (EA Signaling) is available and is subject to availability of the needed facilities (SDN voice or DSA data) for interconnection to AT&T.
 - ◇ *Switched Access Voice:* Customers may use switched access to initiate SDN voice calls by pre-subscribing to AT&T's SDN Carrier Access Code (CAC) of 732 or by dialing a 10732 prefix (indicating an AT&T SDN voice call, if the customer is pre-subscribed to a different CAC). Callers may dial (10732+) 1+NPA-NXX-XXXX, (10732+) 011+CC-NN, or (10732+) 1+700+7D, where 7D is the SDN 7-digit Private Number. (The 10732 prefix is dialed only if the customer is not pre-subscribed to AT&T SDN voice). The dialed digits (specifically, the NPA-NXX-XXXX, 011+CC-NN, or 7D portion of the dialed string) are used to determine the destination and call type in the same manner as for special access-originating SDN calls (see Table I - 2).
 - ◇ *Switched Access Data:* Customers may use switched access to initiate SDDN data calls by pre-subscribing to AT&T's Carrier Access Code (CAC) of 288 or by dialing a 10288 prefix (indicating an AT&T call). Callers may dial (10288+) 1+10D, where 10D is an number of the format NPA-NXX-XXXX or a 10-digit private number (for example, to access AT&T Global Business Video Service). Callers may initiate international (Non-WZ1) data calls by dialing (10288+) 011+CC-NN. The dialed digits (specifically, the 10D number or the 011+CC-NN digits in the dialed string) are used to determine the destination and call type in the same manner as for special access-originating SDDN calls (see Table I - 3). Callers may also dial (10288+) 1+700-7D, in which case the 700-7D is processed as an SDDN 700-Outdialing call whenever the 700-7D number has the format 700-56X-XXXX or 700-73X-XXXX. (As used for switched access data calls, the 7 digits following the 700 code are processed as a 7-digit Private Number only when the string is not of the format 56X-XXXX or 73X-XXXX.)
 - ◇ *Switched Access Voice and Data:* Customers may use switched access to initiate both voice and data SDN calls, subject to the conditions above. Note that, under this arrangement, customers can only use pre-subscription for one type of call (voice or data); the customer is required to dial a prefix for the non pre-subscribed call type (10288 for data or 10732 for SDN voice).
 - The ACP will output 0 to 7 address digits to special egress CPE. When ordering service, the customer must choose how many digits are to be outputted, on a per service, per trunk subgroup basis.

Additional dialing plan information may be found in the AT&T Business Communications Services Guide.

4.1.3 SDDN Access Options

Currently, there are seven ways customers access SDDN: three types of special access, and four types of switched access. Complete interworking is supported for SDDN 56 Kbps service, i.e., there are no restrictions regarding symmetry of access arrangements. The SDDN 64, 384 and 1536 Kbps services require end-to-end ISDN connections.

Figure I - 1 shows three types of special access to SDDN: DDS, T1.5² and PRI. The specifications for DDS access are contained in TRs 62310^[6] and 41458. The signaling specifications for T1.5 are contained in TR 41458; the specifications for PRI are contained in TR 41459. The SDDN data rate must be 56 Kbps if DDS or T1.5 access is used. The SDDN data rate may be 56, 64 384 or 1536 Kbps if PRI access is used.

Two provisioning options are available for T1.5 access - static and dynamic. Static access means that the customer specifies that a certain number of the 24 channels on the T1.5 access interface are dedicated to SDDN. Dynamic access means that any one of the 24 channels may be used either for voice or data (SDDN). Dynamic access requires that the customer's CPE be able to provide an in-band "signal" to the network that a particular call is a data call. Two techniques have been implemented for dynamic access: feature code dialing and station group designator (SGD). The feature code "115", prefixed to the dialed number, indicates a data call. Note that the 115 is prefixed to a 7-digit dialed number; it is not currently supported on the other SDDN dialing plans, i.e., 10-digit private numbering plan, 10-digit NANP, and CC+NN international numbering plan. The SGD is a one-digit suffix, which is added to a dialed number. Use of an SGD is on a subscription basis.

2. NOTE: Within this document, T1.5 refers to T1.5 Service using Robbed-bit signaling.

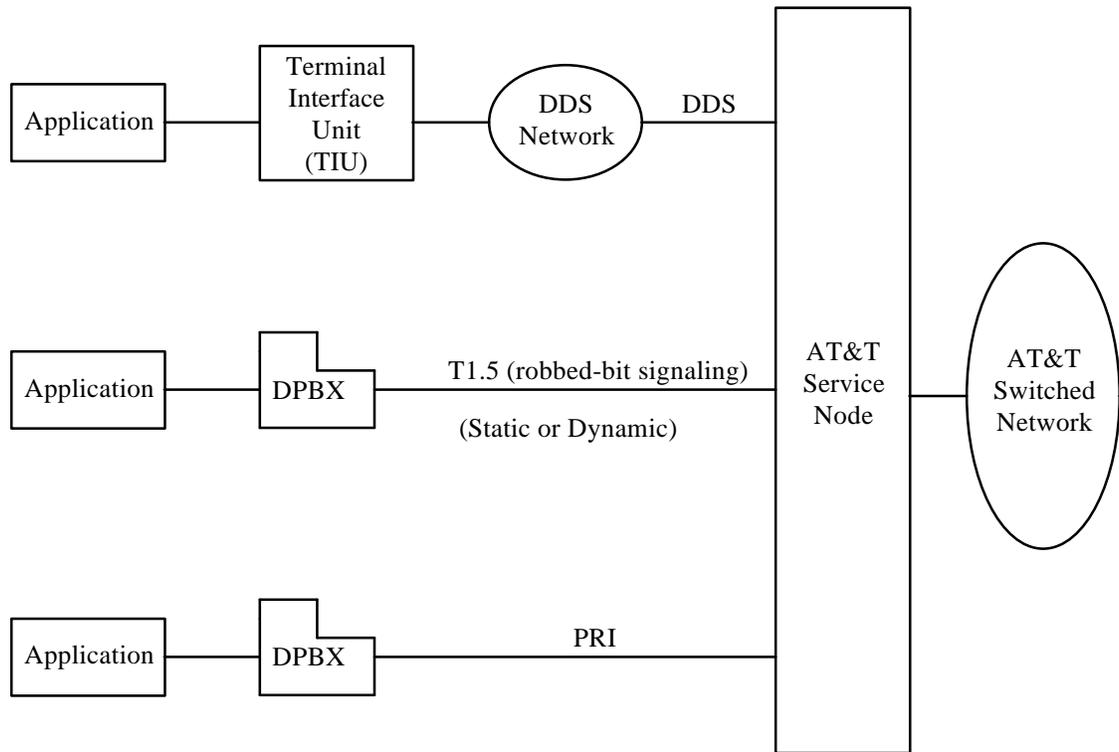


Figure I - 1. Three Types of Special Access to SDDN, SDS, SDI and SDDN-I

Figure I - 2 shows the four types of switched access to SDDN: DDS, Datapath, ISDN Basic Rate Interface (BRI) and ISDN PRI. These access interfaces terminate in a Local Exchange Carrier's (LEC) Central Office (CO). Typically, a LEC tandem office concentrates circuit-switched data traffic from several COs. The traffic is provided to AT&T over FG-D trunks, with either in-band signaling or Network Interconnect (NI) signaling. SDDN 56 Kbps service can be provided using any of the combinations shown in Figure I - 2. Switched access to SDDN 64 Kbps service requires BRI or PRI access, SS7 between LEC switches, and NI signaling between the LEC switch and the AT&T POP.

Note that DDS special access is provided from AT&T, and DDS switched access is provided from the LECs. Although the same four-wire technology is used, different signaling protocols may be required. See Bellcore's TR-EOP-000277 for Datapath specifications.

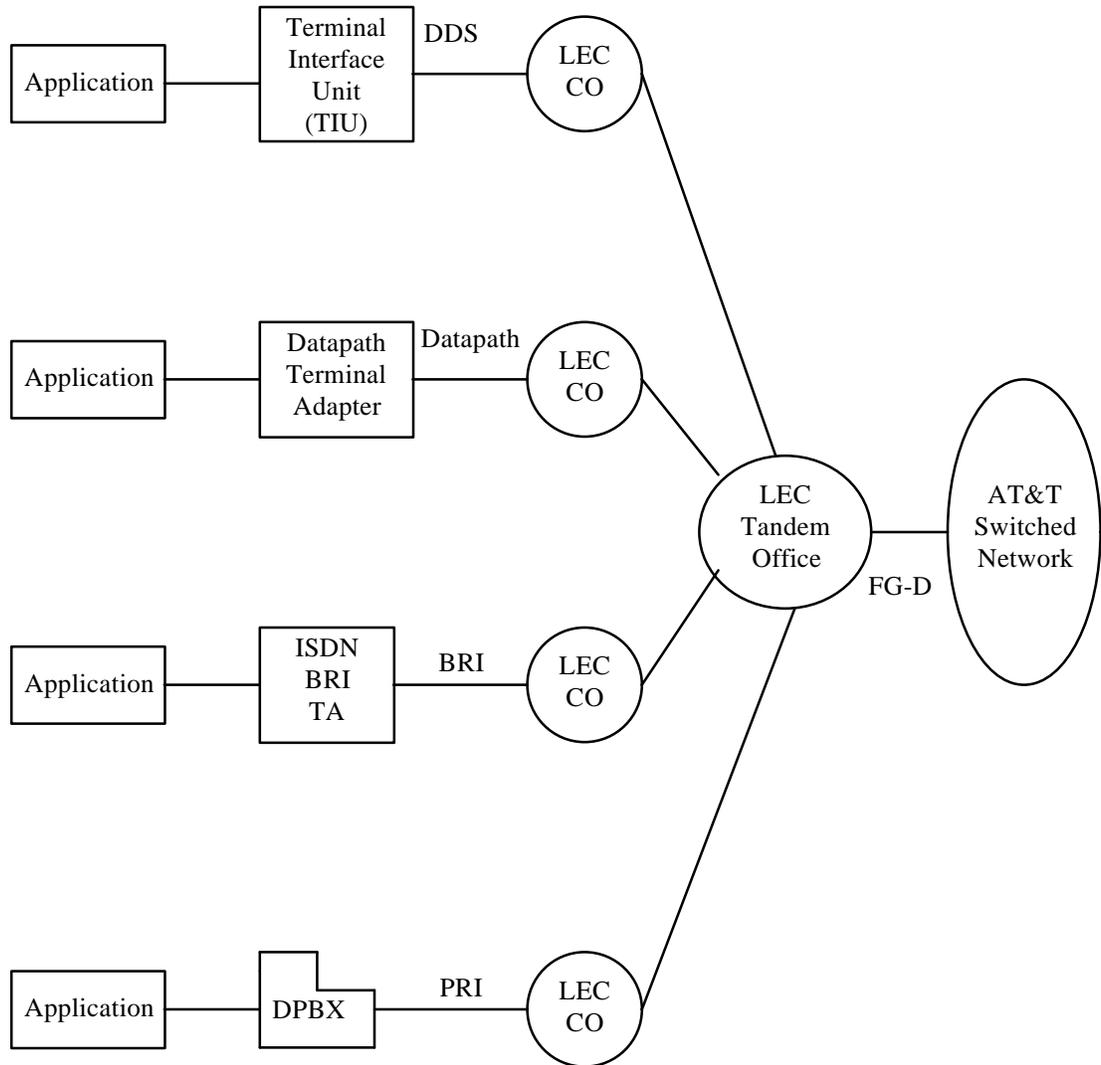


Figure I - 2. Four Types of Switched Access to SDDN, SDS, SDI and SDDN-I

4.1.4 SDDN Off-Net Options

Figure I - 4 describes the SDDN 700 In-Dial feature, which allows customer locations with SDDN special access to receive off-net originating calls. These calls may be originated by SDS customers, other SDDN customers, or international customers using AT&T SDI Inbound service.

Part I - General Information

SDDN customers who subscribe to the 700 In-Dial feature will have 700 numbers assigned to their on-net special access SDDN locations. Customers with 700 In-Dialing may receive calls originating from any or all of the the off-net switched digital access options.

1. The caller may have one of the three types of special access (DDS, T1.5 or PRI).
2. The caller may have one of the four types of switched access (DDS, Datapath, BRI or PRI).
3. The call may be an inbound international call.

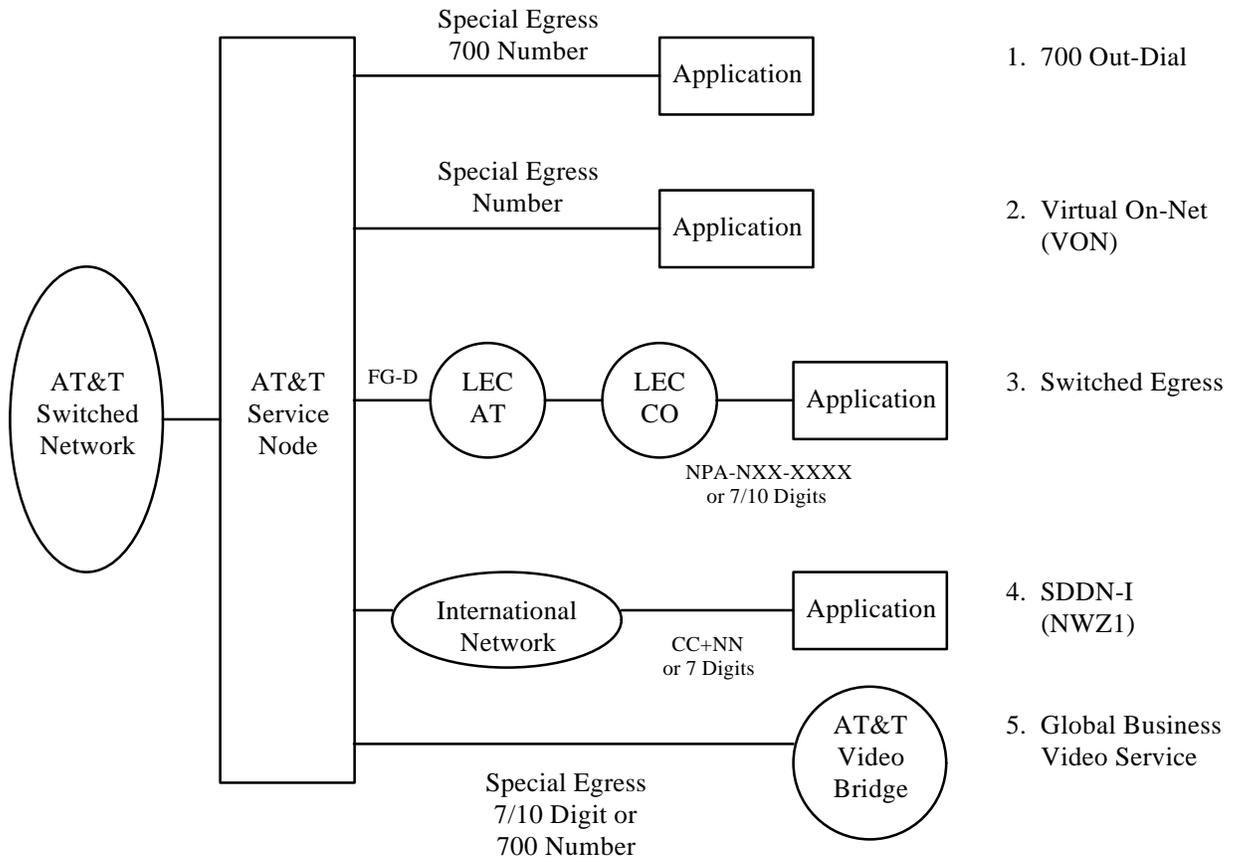


Figure I - 3. Five SDDN Off-Net Egress Options.

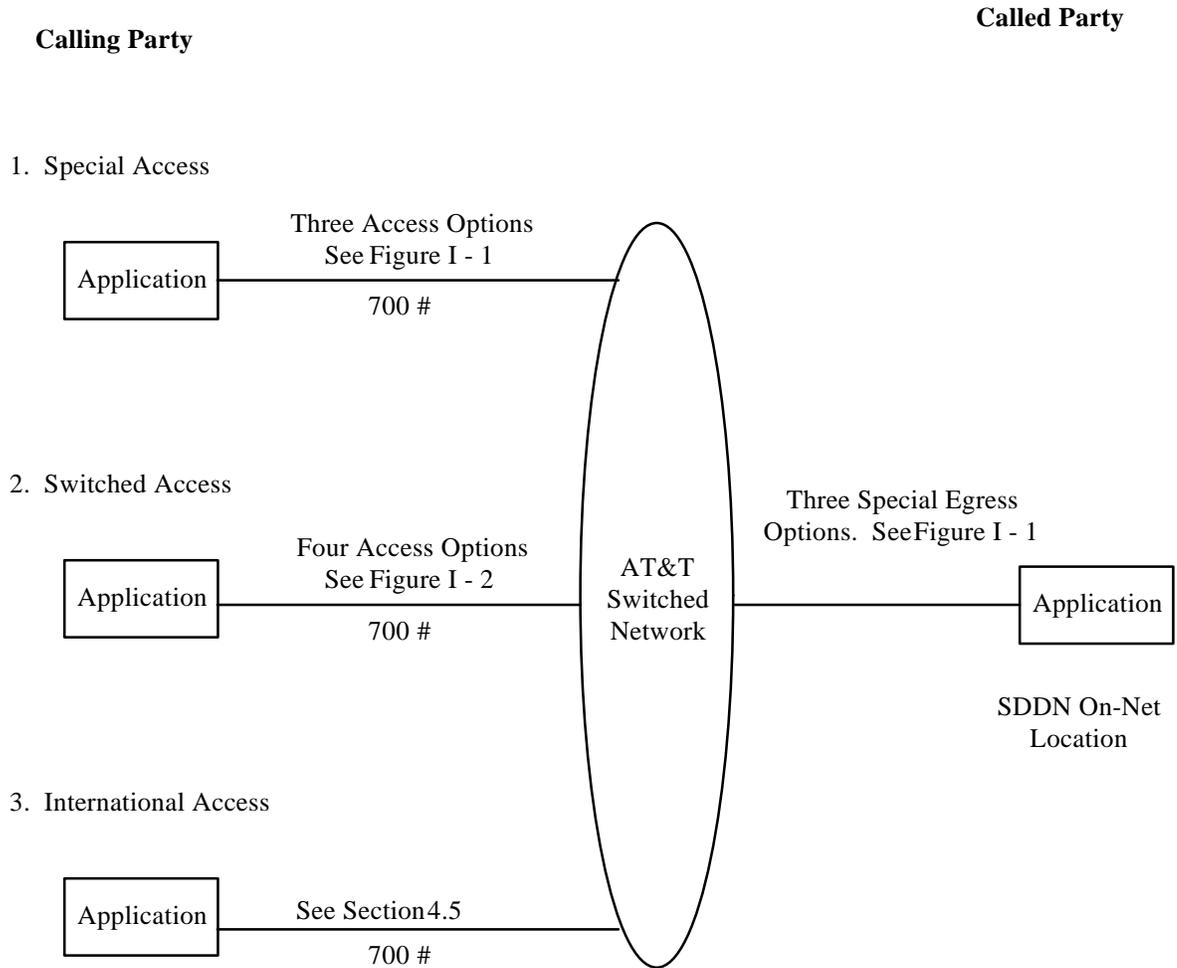


Figure I - 4. SDDN 700 In-Dial Feature

4.2 MEGACOM[®]

MEGACOM is a nodal-based, outward calling, ubiquitous termination service that uses the AT&T switched network to transmit voice and voiceband data communication. MEGACOM[®] offers customers a calling capability similar to that of Outward Wide Area Telecommunications Service (OUTWATS), with no band restrictions nor band pricing. Calls may be completed to the contiguous United States, Alaska, Hawaii, Puerto Rico and the U.S. Virgin Islands. Intra-state, Canada, and international service is available as an add-on. Special access as specified in Part II of this publication is required between the AT&T Service Node and the customer's location. Access to MEGACOM[®] may be via the two-wire loop or ground start interface (for voice-only applications), the four-wire E&M interface, or the four-wire E&M provisioned over the DS-1 access. Voiceband data applications require either the four-wire E&M interface or its equivalent provisioned over the DS-1 access facility. Access may also be via the AT&T ISDN Primary Rate Interface.

4.2.1 Special Features

MEGACOM[®] customers may also subscribe to the following optional features:

- **Call Detail Billing/Bill Analysis** - gives customers a listing of the individual calls made on their trunk subgroups providing called number, time, date, and duration of the calls. Call charges as well as total monthly charges will be recorded.
- **International MEGACOM[®] Add-On** - allows an interstate MEGACOM[®] customer to reach all dialable international dialable countries/points.
- **Canadian MEGACOM[®] Add-On** - allows an interstate MEGACOM[®] customer to reach all dialable Canadian NPAs.
- **Intra-State MEGACOM[®] Add-On** - offered only on a full-state basis in states that have approved intra-state MEGACOM[®].

Additional features and detailed information may be found in the AT&T Business Communications Services Guide.

4.2.2 Dialing Plans

- A special access MEGACOM[®] customer can call switched access locations in the United States and Canada, as well as United States and non-United States territories of the Caribbean using a number of the NPA-NXX-XXXX format. Note that there is no 1+ dialing on MEGACOM[®].
- A MEGACOM[®] customer can call an international location (non-world Zone 1) using a number of the 011+CC+NN format, where the 011 is the International Long Distance Service (ILDS) prefix.

4.3 Toll Free MEGACOM[®]

All Toll Free services are designed to allow a customer to receive telephone calls originated within prescribed service areas without a charge to the calling party. Toll Free MEGACOM[®] (formerly referred to as MEGACOM[®] 800) is offered to customers who receive a large number of calls of a fairly long duration. With the basic service, Toll Free MEGACOM[®] customers may receive calls from all service areas (NPAs).

For states with an intrastate tariff in effect, home state NPAs are included. In addition, a customer may use the Customer Selected Service Area/NPA (CSSA) feature to receive toll free calls only from selected service areas or NPAs. Calls from excluded service areas/NPAs will be screened and blocked. Special access, as specified in Part II of this Publication, is required between the AT&T Service Node and the customer's location. Access may be 4-wire E&M analog, T1.5 using robbed-bit signaling, ISDN PRI, two-wire ground start, two-wire loop start or and two-wire loop reverse battery.

4.3.1 Special Features

ANI Delivery - this feature allows specific calling and called party information to be passed to the terminating CPE over Nodal Dual-Tone Multi-Frequency (DTMF) trunks. For this feature, the calling party information passed to the CPE is the originating caller's Billing Number (BN), also referred to as ANI (Automatic Number Identification).

For switched access calls, the BN is the ANI received from the Local Exchange Carrier (LEC) via Equal Access MF signaling, the number in the Charge Number parameter received via SS7 Network Interconnect ISDN User Part (ISUP) signaling, or the number contained in the Calling Party Number (CPN) parameter received via SS7 NI ISUP signaling if only the CPN and Originating Line Information parameters are received. For direct access (nodal) calls, the BN is stored within the AT&T network.

This terminating feature is supported on Toll Free MEGACOM®, MultiQuest® and Software Defined Network (SDN) nodal services using DTMF Wink Start signaling. Details concerning nodal (T1.5) access and DTMF Wink Start signaling to the AT&T network can be found in AT&T Technical References TR 6241¹⁷ and TR 41458 respectively.

To support this feature, CPE should be capable of extracting the ANI and DNIS (Dialed Number Identification Service) digits from the DTMF signaling stream if not already capable of doing so. The ANI digits will generally be of the NANP format, but may in some cases consist of less than 10 digits as a function of the signaling type used to originally deliver the call to the AT&T network.

Delivery of the call to the terminating CPE will follow the existing procedures and timing intervals for DTMF Wink Start signaling with the following exceptions: Following outgoing trunk seizure by the network and "wink" from the CPE, the network will begin the outpulsing of digits in accordance with existing DTMF digit generation and timing intervals. The digits outpulsed by the network will be of the format:

- a) *ANI*DNIS*
- b) **DNIS*

Where:

* is the asterisk digit in DTMF signaling, used as a delimiter,

ANI is the calling party's BN digits,

DNIS is the Dialed Number Information Service digits.

In case a) the calling party's ANI is available from the network.

In case b) the calling party's ANI is not available from the network.

Once the digit outpulsing is complete, normal Wink Start DTMF call processing shall resume.

AT&T provides a wide range of additional features available to Toll Free MEGACOM customers. A detailed listing of all features available may be found in the AT&T Business Communications Services Guide.

4.3.2 Dialing Plans

- A Toll Free MEGACOM® customer can be reached only from switched access locations using a number of the 8YY-NXX-XXXX format.
- The ACP will outpulse 0 to 7 address digits to egress CPE. When ordering service, the customer must choose how many digits are to be outpulsed, on a per service, per trunk subgroup basis.

4.4 Switched Digital Services (SDS)

4.4.1 Switched Digital 56 Service

Switched Digital 56 Service (SDS 56) is an end-to-end digital service, providing 56 Kbps synchronous, circuit switched channels. Except for the Digital Broadcast Capability feature (Section 4.4.5), the service is full duplex.

The same network platform is used to provide SDDN, Toll Free Multimedia and the Switched Digital Services. This platform consists of AT&T network switches, interoffice facilities (100% terrestrial fiber), SS7 signaling nodes and network databases. The interoffice facilities are equipped with a per-call echo cancellation capability, therefore it is not a requirement that CPE disable echo cancellers in the AT&T network.

The three types of special access shown in Figure I - 1 and the four types of switched access shown in Figure I - 2 apply to SDS 56 as well as SDDN. Note that the ISDN access types, BRI and PRI, are capable of 56 and 64 Kbps transmission. See section 4.4.6 for dialing plan information.

Inverse multiplexers are used to establish nx56 connections by dialing multiple SDS 56 calls. For example, Figure I - 5 shows a 336 Kbps connection for video conferencing. In this example, the AT&T network establishes six separate SDS 56 calls. These calls may take different physical paths, so it is a requirement that the inverse multiplexers provide buffering and synchronization to compensate for differential propagation delay. Inverse multiplexers can also be used to establish Nx64 and Nx384 connections.

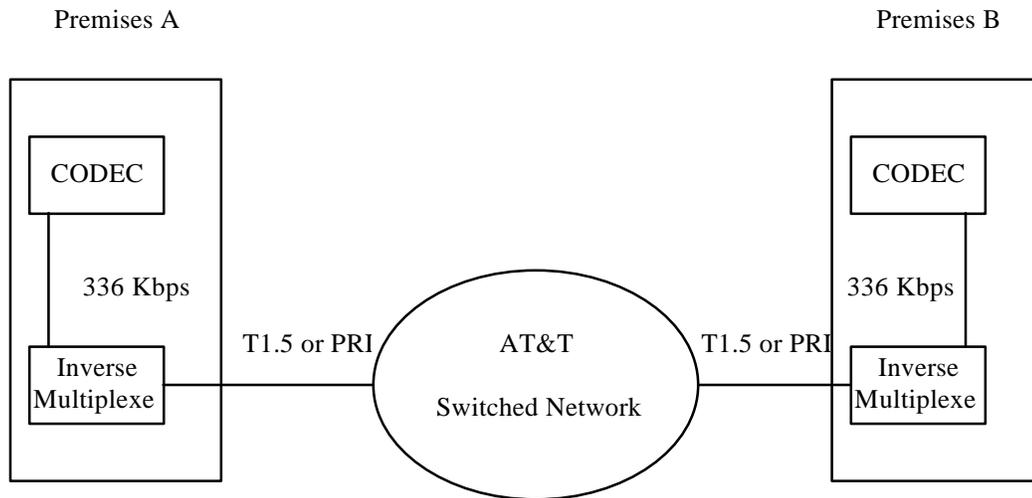


Figure I - 5. Inverse Multiplexing - Video Application

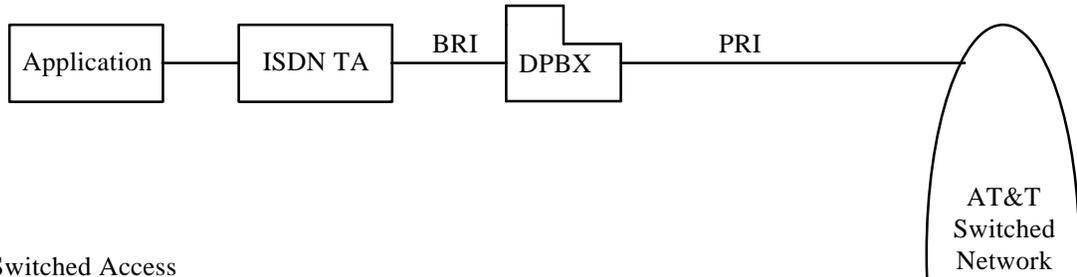
Six 56 Kbps calls established, resulting in a 336 Kbps video connection.

4.4.2 Switched Digital 64 Service

Switched Digital 64 Service (SDS 64) is an end-to-end digital service, providing 64 Kbps synchronous, full duplex, circuit switched channels. SDS 64 is provided on the same platform as SDS 56. SDS 64 requires an end-to-end 64 Kbps synchronous connection. Figure I - 6 shows two types of ISDN access currently available, BRI and PRI. Note that for switched access to SDS 64, Network Interconnect signaling must be available between the LEC and AT&T.

The AT&T network does not provide automatic rate adaptation, e.g., fall-back from 64 to 56 Kbps if the egress does not support 64 Kbps. It is a requirement on the originating station to correctly specify the Bearer Capability Information Element to the AT&T network. Thus, if the egress connection is other than those shown in Figure I - 6, the originating station should specify 56 Kbps not 64 Kbps in the Bearer Capability Information Element. In the near future, call originations of 64 Kbps and higher will not be delivered from the network to 56 Kbps special access locations. Sub-rate adaptation other than 56 Kbps is not explicitly supported by AT&T and will be processed as a 64 Kbps call.

1. Special Access



2. Switched Access

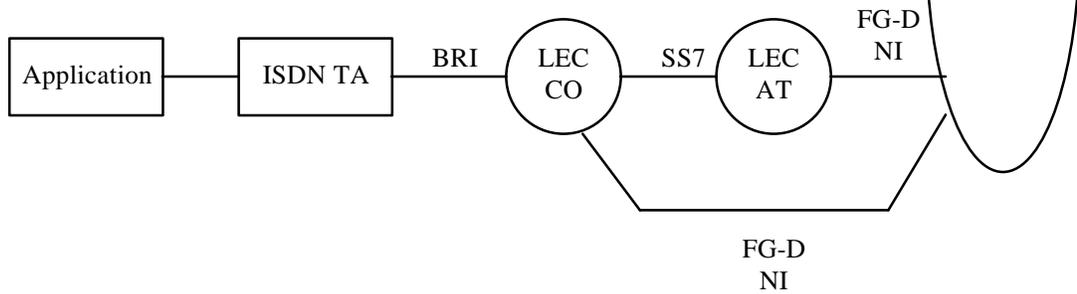


Figure I - 6. Two Types of ISDN Access to Switched Digital 64 Service (BRI and PRI).

4.4.3 Switched Digital 384 Service

Switched Digital 384 Service (SDS 384) is an end-to-end digital service, providing 384 Kbps synchronous, full duplex, circuit switched channels. Currently ISDN PRI access (see Figure I - 1) is a requirement for SDS 384.

Switched Digital 384 channels are referred to as H0 channels in ISDN terminology. SDS 384 requires that the CPE allocate 384 Kbps calls on contiguous channels, with specific time slots. The required time slots are 1-6, 7-12, or 13-18. Channels 19-24 may be used if the access facility is Non-Facility Associated Signaled (NFAS), i.e., the signaling for the channels on this T1.5 interface is provided by another PRI access connecting the CPE to the AT&T network (see Figure I - 7).

4.4.4 Switched Digital 1536 Service

Switched Digital 1536 Service (SDS 1536) is an end-to-end digital service, providing 1536 Kbps synchronous, full duplex, circuit switched channels. Switched Digital 1536 channels are referred to as H11 channels in ISDN terminology. As shown in Figure I - 7, two T1.5 interfaces are required to establish an SDS 1536 call. The first T1.5 contains the signaling channel (D-Channel). The second T1.5 carries the 1536 channel. The SDS 1536 call requires all 24 channels on a single NFAS facility, starting with channel 1.

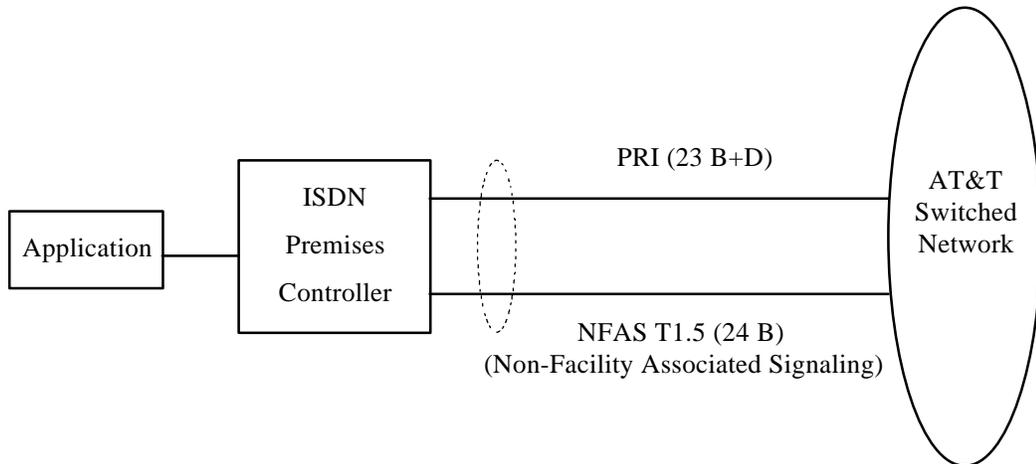


Figure I - 7. Access to Switched Digital 1536 Service

4.4.5 Digital Broadcast Capability

The Digital Broadcast Capability (DBC) is an AT&T network-provided half-duplex bridge supporting 56 and 112 Kbps transmission from the broadcasting leg to the receive-only legs on the conference. The DBC is a “Meet-Me” bridge, and conferences must be reserved. Two 700-56X-XXXX numbers are provided when the reservation is made, one for the originator, the other for the receive-only conferees. No calls are originated by the DBC.

Access to the DBC, at 56 Kbps, can be made from any of the three types of special access shown in Figure I - 1 and the four types of switched access shown in Figure I - 2.

4.4.6 Switched Digital Services Dialing Plan

Figure I - 8 shows the dialing plan for the Switched Digital Services. Three scenarios are shown: switched egress, ISDN PRI egress, and DDS/T1.5 special egress.

Switched egress calls are of the format NPA-NXX-XXXX and are assigned by the LEC. Switched egress calls may be 56 or 64 Kbps. Note that Network Interconnect signaling between the LEC and AT&T, and either PRI or BRI access is required for 64 Kbps.

ISDN PRI, a special egress option, provides SDS 56, 64, 384 and 1536 services. The caller dials 700-73X-XXXX to reach a station behind an AT&T PRI interface.

DDS and T1.5 are special egress options, providing egress to SDS 56 service. The caller dials 700-56X-XXXX to reach a station behind a DDS or T1.5 egress line.

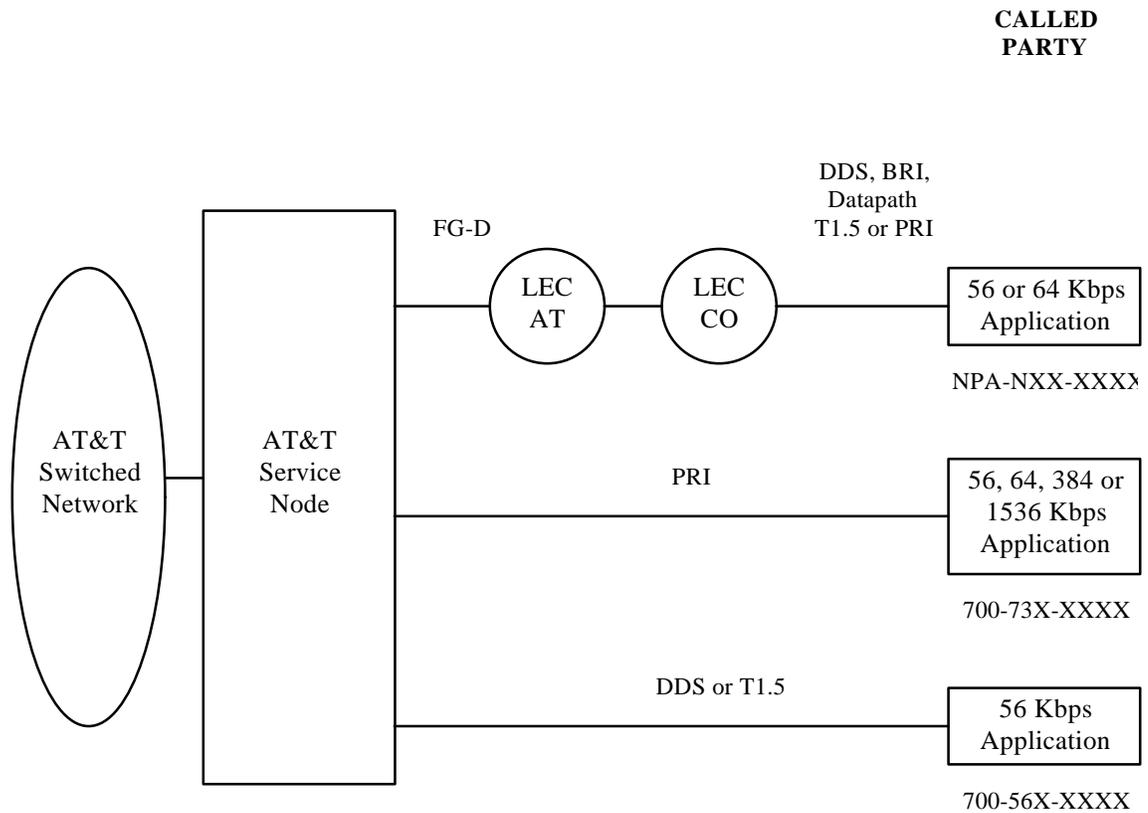


Figure I - 8. Switched Digital Services Dialing Plan

4.5 Switched Digital International (SDI)

SDI and SDDN-I complement the two domestic switched digital service families, Switched Digital Services (SDS) and SDDN, respectively. SDI and SDDN-I are offered on the same transport network platform. Any SDS customer may place an SDI call; only SDDN customers may place SDDN-I calls.

SDI and SDDN-I are currently available to numerous countries including some countries with multiple carriers. Additional information may be found in the AT&T Business Communications Services Guide.

SDI and SDDN-I currently support 56 and 64 Kbps transport rates. SDI / SDDN-I 384 Kbps service is supported to a limited number of countries. Support of SDI / SDDN-I 1536 Kbps service is planned. The access arrangements shown in Figure I - 1 and Figure I - 2 apply to SDI and SDDN-I. Inverse multiplexer (see Figure I - 5) can be used on SDI and SDDN-I calls to provide nx56 and nx64 connections.

4.5.1 Inbound SDI Calls

The dialing plan for inbound SDI calls is the same as SDS, shown in Figure I - 8, with one exception.

The exception is for the following scenario:

- the inbound SDI call is to a station behind an ISDN PRI interface,
- the inbound SDI call is 56 Kbps, not 64 Kbps, and
- the signaling on the international link uses CCITT No. 7 TUP (Telephone User Part), not ISUP (ISDN User Part).

For this scenario, the dialed number is not 700-73X-XXXX as shown in Figure I - 8, but rather 700-56X-XXXX, 700-556-XXXX or 700-956-XXXX. Additional codes will be added as needed.

4.5.2 Outbound SDI Calls

The dialing plan for outbound SDI calls is:

Special access (Figure I - 1)

- ISDN PRI: 011+CC+NN
- DDS and T1.5: 173+CC+NN³

Switched access (Figure I - 2): 011+CC+NN

4.5.3 SDDN-I Calls

Currently, international on-net to on-net connectivity for data traffic is not supported by AT&T (it is planned for introduction as part of a Global Software Defined Data Network -- GSDDN -- service offering).

3. The 173 access code used with DDS and T1.5 will be phased out in favor of the 011 access code.

Domestic SDDN customers who wish to receive inbound international calls may do so by subscribing to the SDDN 700 In-Dial capability (Figure I - 4), described in Section 4.1.4.

The dialing plan for outbound SDDN-I calls (on-net to off-net, Figure I - 3) is defined in Section 4.1.2 and in AT&T TR41459 (ISDN PRI).

4.6 Alternate Egress Arrangement (AEA) For Long Distance Service

AEA will terminate ASN calls directly to the customer premises. It will be provisioned on an individual case basis at locations with sufficient traffic volumes over dedicated facilities arranged by the customer. It may be part of a package offered with other AT&T communications services.

AEA will redirect inter-LATA ASN switched traffic which is billed to both the call originators and the call receivers. Calls may originate over many AT&T services, such as DDD, AT&T Card, WATS, MEGACOM, CNO Off-Net, CCSA Off-Net, SDN Off-Net, IDDD, and other services which are billed to the call originator. Any other AT&T Inbound Service which is normally served by the LEC Class 5 Office at the terminating location will also be included. AEA will not impact the rates charged for these services.

Services requiring special access facilities, such as Toll Free MEGACOM and on-net traffic cannot be included in an AEA.

The provider of each AEA must commit to furnish a quality of service that meets the performance requirements of the ASN. This quality of service must include service availability, mean time between failure, mean time to repair transmission parameters, and accuracy of usage information.

AEA will be provided where the customer has the complete range of stations behind the d-digits (thousands series). Calls may be screened down to the tens group of numbers. The customer will receive the complete 7-digit message network number and will be fully responsible for switching the call to the appropriate station.

AT&T will implement AEA at locations:

- Which attract sufficient calling volumes
- Where the customer has a complete thousands series (or multiples thereof) range of stations.
- Where the customer has deployed CPE which AT&T has approved for interconnection as a switching device of the ASN. The CPE must provide the same functionality for answer supervision and signaling that is provided by the LEC Class 5 Office in the traditional method of switched traffic egress. Customer locations served by a CENTREX may be included in an AEA.

4.7 MultiQuest[®] 900 Service

The AT&T MultiQuest[®] (MQ) Service provides 900-inward calling to customers with direct egress from an AT&T Service Node. This domestic service is targeted for high volume inward calling terminating at the sponsor's location.

MultiQuest[®] was introduced by AT&T to allow our customers (sponsors) to offer a valued service to their clients (callers) and charge premium rates. Sponsors will choose the premium charge they wish to apply under a Premium flexible billing agreement.

Callers can dial a 900 number, complete to customer's premises multiplexing equipment over T1.5 egress facilities and communicate with live attendants, voice messaging equipment, and computer databases to

obtain value-added services. This interactive capability distinguishes MultiQuest® from DIAL-IT 900 Service. Callers pay for the customer's service via a premium charge on their telephone bill. These charges are forwarded to the customers by AT&T after certain transport and contract deductions have been made.

MultiQuest® customers must comply with the Federal Trade Commission (FTC) rules pursuant to the Telephone Disclosure and Dispute Resolution Act (TDDRA), (FTC file number R311001, 900 Pay-Per-Call).

4.7.1 Special Features

- Call Detail Report - Bill analysis gives the customer, at no charge, a list of individual calls on their TSGs with complete call details. Call charges and total monthly charges will also appear.
- Executive Summary Report - A monthly report that customers will receive, at no charge, for the prior billing period, which provides summary usage information by NPA for each AT&T MQ 900 number.
- Flexible Billing - Allows a sponsor to charge any rate that they want, subject to AT&T's parameters, on a per 900 number basis. The sponsor will no longer use a fixed rate associated with a 900 number. If the sponsor wishes to change the rate, the 900 number can remain the same .
- Electronic Funds - Allows a sponsor to receive premiums directly into their account via an AT&T transfer procedure.
- ANI Delivery (see Section 4.3.1)

The Call Management Features for MQ are identical to the Advanced 800 FCC Tariff No. 2 Offerings, with the exception of Courtesy Response. Additional features and information may be found in the AT&T Business Communications Services Guide.

4.7.2 Dialing Plan

Callers will originate a call to a sponsor by dialing "1-900-NXX-XXXX".

- MQ uses a SAC, which is a subset of the NANP standard (1-900-NXX-XXXX).
- Dialable numbers are allocated to carriers on a six digit basis (900-NXX).
- Originating calls are switched through the LEC office.

4.8 M44X Multiplexing

M44X is a multiplexing service function of T1.5 service that is fully described in AT&T Technical Reference PUB 54070. The M44X multiplexing service function typically allows a maximum of 44 32Kbps channels to be transmitted on a single 1.544 Mbps transmission facility .

M44X, using ADPCM devices such as the BCM 32000X, will ONLY be allowed on circuit configurations for MEGACOM®, Toll Free MEGACOM®, and MultiQuest® Service. 9.6 Kbps analog data will not be supported. 4.8 Kbps and voice applications will be fully supported.

When BCM 32000X is used and the round trip delay on the access facility exceeds 11 ms, it may be necessary to have the customer to provide echo cancelers on the non-compressed inputs of the M44X multiplexing device at the customer premises.

4.9 Toll Free Multimedia

AT&T's Toll Free Multimedia Service allows customers to receive data and voice calls on the same toll free number. It is a dial-up service for multimedia offerings which are various combinations of video, voice and data on a single connection. The service will support originating access by:

- 64 Kbps clear connection on an ISDN BRI ordered from the Local Exchange Carrier, where available,
- any form of switched 56 Kbps service (e.g., Switched Digital 56 Service) tested by the LEC for use with the Toll Free Multimedia Service,
- voice (see Section 4.3),
- 384 Kbps connection on an ISDN Primary Rate Interface (planned).

Subscribers must connect to the network via ISDN PRI at the terminating location. Additional information may be found in the AT&T Business Communications Services Guide.

4.9.1 Dialing Plans

AT&T's Toll Free Multimedia Service customers can be reached from switched access locations using a number of the 8YY-NXX-XXXX format.

4.10 Universal T1.5 Access

Universal T1.5 Access allows multiple services to be carried over a single T1.5 using either DP, DTMF, MF or ISDN PRI signaling.

The capabilities and services that can be carried on a Universal T1.5 Access trunk include:

- Access Services
 - ◊ MEGACOM[®]
 - ◊ Operator requested calls
- Egress
 - ◊ Toll Free MEGACOM[®]
 - ◊ Long Distance Service
- Two-Way Service
 - ◊ SDN
 - ◊ SDS capabilities (planned)

Depending on the signaling associated with the customer's T1.5, it may not be possible to offer all of the capabilities on the same trunk. Only ISDN PRI trunks are capable of supporting more than one 1+ voice

service on access; all other arrangements must choose between SDN or MEGACOM[®]. In addition, DP, MF and DTMF trunks are unable to support data rates in excess of 56 Kbps.

4.10.1 Dialing Plans

The dialing plans for the respective services shall apply to Universal T1.5 Access. Customers can reach AT&T Operator Services by using numbers of the following format:

- 00
- 0-NXX-XXXX
- 0-NPA-NXX-XXXX

International Operator Services may be reached by dialing a number of the 01+CC+NN format.

5. GLOSSARY OF TERMS AND ACRONYMS

Access Line or Trunk: In the context of this document, the term access line or trunk refers to a special access, DS-0 level or voiceband analog channel extending from an ACP to a customer's premises. An access line or trunk may be carried over two-wire ground start, four-wire E&M, or DDS facilities, or may be one of the twenty four DS-0 channels within a DS-1 level facility.

ACP: See Action Point.

Action Point (ACP): An intelligent node at the edge of the AT&T network which accepts user's call requests, and under stored program control can provide the required routing, billing, record keeping, and other necessary functions.

Alternate Mark Inversion (AMI): A digital line coding scheme in which binary ones are represented by pulses which alternate in polarity, and binary zeros are represented by the absence of pulses. Any violation of this polarity alternation may be regarded as an indication of a transmission error, so some types of terminal equipment log bipolar violations for maintenance purposes.

Answer Signal: An off-hook indication transmitted by a called CPE toward the egress ACP to indicate that the call has been answered. If so provisioned, the answer signal propagates back to the calling CPE or LEC switching office.

Audible Ring Tone: A tone transmitted to the calling CPE to indicate that the called CPE is being alerted of the incoming call. Audible ring tone is transmitted over the voice channel of four-wire analog facilities, or is PCM encoded on DS-1 facilities.

AMI: See Alternate Mark Inversion (AMI).

B8ZS: See Bipolar with 8 Zero Substitution (B8ZS).

BER: See Bit Error Rate.

Bipolar with 8 Zero Substitution (B8ZS): A digital line coding scheme which is similar to AMI, but which ensures sufficient pulse density on a DS-1 facility. B8ZS is especially important when transmitting digital data, since DS-1 receiver clock circuits must synchronize to the pulses in the received signal; digital data can contain long strings of zeros, which, if not for the B8ZS scheme, would result in extended periods during which the signal would have no pulses, and therefore be problematic. B8ZS enforces the pulse density constraint by encoding each string of eight consecutive zeros as a special bipolar violation sequence.

Bit Error Rate (BER): A figure of merit for digital communications systems computed by dividing the total number of bits received in error in a given time interval by the total number of bits transmitted in that same time interval.

Channel Bank: Terminal equipment for transmission system used to multiplex individual channels using either Time Division Multiplexing (TDM) or Frequency Division Multiplexing (FDM) techniques for transmission over a higher-bandwidth carrier system. While other types exist, only SF-compatible and ESF-compatible digital channel banks are relevant to this document. This type of channel bank multiplexes 24 lower-bandwidth signals, on a time-division basis, for transmission over a digital carrier system. The lower-bandwidth signals may be encoded analog (e.g., 2-wire ground start, 4-wire E&M, etc.) or digital (e.g., 56 Kbps DDS line coding). The associated channel units perform whatever analog-to-

digital conversion or digital line coding conversion is necessary. The demultiplexing operation is the inverse of the multiplexing operation.

Centrex: A local, LEC-owned switching system which connects subscriber lines and provides PBX-like features.

Class 5 Central Office: A local, LEC-owned switching system, which connects subscriber lines to other subscriber lines or to interoffice trunks.

CPE: Customer Premises Equipment.

CPE&A: Customer Premises Equipment plus the Access facilities, viewed as a single arrangement interfacing to AT&T's nodal services at the AT&T point of termination.

Data Communications Equipment (DCE): Customer premises equipment which adapts digital signals from data terminal equipment to a format suitable for transmission over the network, and which controls the connection.

Data Terminal Equipment (DTE): In the context of this document, customer premises equipment which is a source and/or sink of digital information. DTE usually cannot terminate a subscriber line, access line, or trunk directly, and therefore must connect to the network through data communications equipment (DCE).

DCE: See Data Communications Equipment.

DDSD: See Delay-Dial Start-Dial signaling.

Decode Loss: The level (in dBm) of a 1004 Hz sinusoid which would result if a digital reference signal was decoded by a zero level decoder.

Delay-Dial Start-Dial Signaling (DDSD): A method of engaging a trunk or access line to set up a call, performing an integrity check on it, and establishing a proper time to transmit address information. For example, suppose a call comes into a PBX over a DDSD access line. The ACP alerts the PBX of the call by transmitting off-hook, i.e. by seizing the access line. The PBX transmits off-hook toward the ACP until it is ready to receive DID address information. This is the delay-dial signal. When ready to receive address digits, the PBX transmits on-hook. The off-hook to on-hook transition constitutes the start-dial signal and is analogous to dial tone. The ACP, satisfied that the channel and PBX are operational (since it has received the delay-dial and start-dial signals), transmits the address information.

Dial Pulse Signaling (DP): A means of address signaling accomplished by momentary on-hook excursions or pulses, whereby the number of such pulses corresponds to the value of the digit.

DID: See Direct Inward Dialing.

Digital Milliwatt: A defined, standardized sequence of PCM words, which when decoded through a zero level decoder, produces a 0 dBm, 1000 Hz sinusoid. Because the digital milliwatt is a finite sequence it is convenient in defining the concept of loss between a digital transmission system and an analog system. However, its utility is purely mathematical; it should not be applied to an actual digital transmission system. The recommended test signal for that purpose is the Digital Reference Signal (DRS).

Digital Reference Signal (DRS): A sequence of PCM words, which when decoded by a zero-level decoder, results in a 0 dBm sinusoid with a nominal frequency of 1004 Hz. DRS differs from the digital milliwatt in that (a) the decoded sinusoid will have a slightly different frequency (1004 Hz as opposed to

1000 Hz), and (b) DRS is not a finite, deterministic sequence. DRS is the correct signal for testing and adjusting digital transmission systems.

DL: See Decode Loss.

DP: See Dial Pulse signaling.

Direct Inward Dialing (DID): A feature that permits incoming calls to stations served by a PBX or other CPE to be dialed directly; the call need not go through an attendant.

DS-0: Digital Signal level 0. See also DS-1.

DS-1: In the time division multiplexing hierarchy of the telephone network, DS-1 (Digital Signal level 1) is the initial level of multiplexing. Traditionally, twenty four 64 Kbps DS-0-level channels have been multiplexed up to the 1.544 Mbps DS-1 rate, with each DS-0 level channel having the capability of carrying the digital representation of an analog voice channel or digital data in various formats.

DTE: See Data Terminal Equipment.

DTMF: See Dual-Tone Multifrequency.

Dual-Tone Multifrequency Signaling (DTMF): A means of address signaling that uses a simultaneous combination of one of a lower group of frequencies and one of a higher group of frequencies to represent each digit or character. DTMF is the signaling method used by modern Touch-Tone telephone sets. DTMF is not the same as multifrequency (MF).

EL: See Encode loss.

Encode Loss (EL): A unitless number which relates the equivalent power represented by the digital output signal of an encoder to the analog power level at the input to the encoder.

ESF: See Extended SuperFrame (ESF) framing and formatting.

Extended SuperFrame (ESF) Framing and Formatting: A time division multiplexing standard used on DS-1 level channels to carry 24 DS-0 level channels. In addition to the features available with SF framing and formatting, ESF offers more robbed bit signaling channels (four, as compared with only two available with SF), a 4 Kbps data channel (reserved for remote alarm indication (yellow alarm) transmission and maintenance purposes), and a Cyclic Redundancy Check (CRC) code. Additionally, the ESF standard circumvents certain situations which could lead to false framing capture and false remote alarm indication recognition if SF framing was used. These new features make ESF the preferred standard for new special access installations. See AT&T Technical Reference TR 54016 for details.

Glare: The condition in which call attempts are simultaneously initiated at both ends of the same two-way trunk or access line. Procedures exist for detecting and gracefully resolving glare on wink start and DDS trunk and access lines, but not for immediate start access lines.

ICL: Inserted Connection Loss.

Immediate Start: A method of engaging a trunk or access line in which no start-dial signal is used. The initiating CPE or office simply seizes the trunk or access line and begins sending address digits.

LEC: Local Exchange Carrier.

M24: A multiplexing function, provided at an AT&T serving office, which provides for characterization of T1.5 Service circuit and allows for connection of the channels to the individual switched and non-switched services offered by AT&T. Refer to AT&T Technical Reference TR 62411 for additional information.

MF: See Multifrequency signaling.

Multifrequency (MF): A type of address signaling in which ten decimal digits and five auxiliary signals are each represented by selecting a pair of frequencies out of the following group: 700, 900, 1100, 1300, 1500, and 1700 hertz (HZ). MF is not the same as DTMF.

NPA: See Numbering Plan Area.

Numbering Plan Area (NPA): Often, NPA refers to a geographical division defined by the familiar area code, within which telephone directory numbers are subgroups. Current NPAs are of the form “NXX”, where:

N = any digit 2 through 9

X = any digit 0 through 9

A few NPA codes within the North American Numbering Plan have been assigned for special uses (such as 8YY for Toll Free service) and are known as Service Access Codes (SACs). The 8YY NPA is shared among the service providers for their service offerings. The 700 NPA has been set aside for any inter-exchange carrier to use for any purpose such as identifying special access endpoints. In this document, the term NPA simply implies a dialed number field having the form NXX, and may be either a geographically significant area code or a Service Access Code.

One-Way Incoming Trunk or Access Line: In this document, a trunk or access line which can only be seized for use by the ACP is designated one-way incoming. The term incoming refers (in this document) to the CPE viewpoint. Once a trunk or access line is seized, 2-way transmission may occur; one-way incoming refers only to the direction of call initiation.

One-Way Outgoing Trunk or Access Line: In this document, a trunk or access line which can only be seized for use by the CPE is designated one-way outgoing. The term outgoing refers (in this document) to the CPE viewpoint. Once a trunk or access line is seized, 2-way transmission may occur; one-way outgoing refers only to the direction of call initiation .

PBX: See Private Branch Exchange.

Private Branch Exchange (PBX): A private switching system, either manual or automatic, usually serving an organization such as a business or a government agency, and usually located on the customer’s premises.

Robbed Bit Signaling: A supervisory signaling scheme used in SF and other DS-1 framing and formatting standards, in which the least significant bit of a channel’s time slot is preempted in selected frames. In SF framing and formatting, two robbed bit channels are available. These are sometimes referred to as signaling channels A and B. In ESF framing and formatting, there are four robbed bit channels, labeled A, B, C, and D. When robbed bit signaling is used in digital data channels such as are used in DS-1 special access to Switched Digital 56 Service, the bit position where bits may be robbed (i.e., the least significant bit) is not available for user data. See also Supervisory signaling.

SF: See Superframe.

Signaling Bit: See robbed bit signaling.

Special Access: Connection of CPE to the AT&T network using dedicated special access service. Although the special access facilities are normally obtained from the LEC (if the CPE and AT&T serving office are in the same LATA), no LEC switching equipment is employed .

Station: In this document, the customer premises equipment which is the endpoint of a call. A station could be, for example, a telephone set or DTE and associated DCE behind a digital PBX served by special access, or a DTE and associated TIU served by a single channel special access line.

Subscriber Line: The facility which extends from a LEC class 5 central office to a customer premises and connects to the customer's station equipment to the LEC switched network.

Superframe (D3/D4) Framing and Formatting: A time division multiplexing standard used on DS-1 level channels to carry 24 DS-0 level channels. Formatting, or channelization format, refers to the sequencing of the 24 time-slots and allocation of the time-slots to their respective DS-0 channels. Framing refers to a synchronization pattern used to establish the time-slot. See AT&T Technical Reference TR 62411 for details of Superframe (SF) and Extended Super Frame (ESF) formats.

Supervisory Signaling: Signaling used to indicate or control the states (e.g. on-hook, off-hook) of circuits involved in a particular connection.

T1.5 Service: A two-point, dedicated, high capacity, digital service provided on terrestrial digital facilities capable of transmitting 1.544 Mbps. Refer to AT&T Technical Reference TR 62411 for additional information on T1.5 service and its functionality.

Terminal Interface Unit: Adapts the DDS line coding to a standard DCE/DTE interface (such as V.35 or RS-449/422).

TIU: See Terminal Interface Unit.

Touch-Tone: See Dual-Tone Multifrequency.

Trunk: A communication channel between two switching systems. See also Access line or trunk.

Two-Way Trunk or Access Line: A trunk or access line that can be seized for use by equipment at either end of the trunk or access line.

Wink Start: A method of engaging a trunk or access line to set up a call, performing an integrity check on it, and establishing a proper time to transmit address information. At first glance, wink start may be viewed as a special case of DDS, where the delay-dial signal is of a short, specified duration. (There is, however, a difference between the glare resolution timing used in wink start and that used in DDS.)

ZCS: See Zero Code Suppression (ZCS).

Zero Code Suppression: A method of providing pulse density enforcement which changes the second least significant bit in any all-zeros octet to a binary one. The ZCS mechanism acts on a per-channel basis, so that any alteration of user information is localized to the particular timeslot in which pulse density would otherwise be low. (This per-channel feature of ZCS contrasts with the method of pulse density enforcement used by several channel service units to provide average pulse density requirements; the latter method typically disregards timeslot boundaries).

Zero Level Decoder: A hypothetical PCM decoder, which accepts 8-bit 11-255 PCM words at its input, and interpolates and expands the samples into an analog output signal. Given a digital milliwatt as its input, the zero level decoder output will be a 0 dBm, 1000 Hz sinusoid .

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-
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 - 3 AT&T Integrated Services Digital Network (ISDN) Primary Rate Interface and Special Application Specification User-Network. Interface Description, Technical Reference TR 41459, August 1996.
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 - 5 AT&T Business Communications Services Guide, Document Number 015-358-027, June 1995.
 - 6 AT&T Digital Data System Channel Interface Specification, Technical Reference 62310, November 1987.
 - 7 AT&T ACCUNET T1.5 Service Description and Interface Specification, Technical Reference 62411, December 1990.

Special Access Connections

To The

AT&T Network

Part II

Special Access Interconnection

At The

AT&T Point Of Termination

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1. SPECIAL ACCESS ELECTRICAL AND LOGICAL INTERFACES

This section lists the voice and data transmission channels and the trunk supervision signaling mechanisms used on the various special access facilities which connect to an ACP. Except for the 56 Kbps single channel access arrangement, these are standard telephony interfaces. The reader should refer to Table I - 1 in Part I, which lists availability by service, of each of these interfaces. Figure II - 1 shows several examples of the special access interfaces to an ACP.

Note: This discussion applies to signaling dialog at the AT&T point of termination. Accordingly, the customer premises equipment and the access facilities (CPE&A) which may or may not be coordinated through AT&T are viewed as a single entity. It is the responsibility of the CPE vendor to interpret the implications of this discussion on equipment design requirements. In installations employing total or coordinated access, AT&T will engineer the access facility requirements. Otherwise, access facility engineering will be the customer's responsibility.

Tolerances for signals detected by AT&T equipment reflect worst case limits. Dispersion, band-limiting, and other distortions may be impressed on the signal during its propagation over the access facility. CPE vendors should allow for these distortions. (ANSI recommendation EIA/TIA-464B provides conservative CPE design criteria with the appropriate margins built in.)

1.1 Two-Wire Lines and Trunks

Two-wire special interface access for lines and trunks is available only on the selected AT&T nodal services indicated in Part I, Table I-1. Moreover, voiceband data users should consider choosing a four-wire interface over a two-wire loop start or ground start interface, since the transmission quality of four-wire access facilities is generally superior to that of two-wire access facilities.

1.1.1 2-wire Loop Start (2LS) and 2-wire Ground Start (2GS)

Special access loop start and ground start interfaces support one-way incoming, one-way outgoing and two-way trunks. Since glare can occur on 2-way trunks, and since 2LS provides no indication of glare to the CPE&A, 2LS signaling is not recommended for 2-way trunks.

The following characteristics apply to both the special access loop start and ground start interfaces:

1. In the inward direction (i.e., towards the CPE&A), trunks are attendant-handled; Direct Inward Dialing is not available.
2. In the outward direction, trunks always use Direct Outward Dialing.
3. Dial tone is provided by the network.
4. The network provides 20 Hz power ringing current to the called CPE&A.
5. Either dial pulse or DTMF address signaling is accepted.
6. Electrical characteristics of both interfaces are typical of industry-standard, two-wire loop start and ground start interfaces.

1.1.2 2-wire loop Reverse-Battery (2RB)

The special access loop reverse-battery interface is typical of industry-standard loop reverse-battery interfaces. It is used for one-way incoming (to CPE&A) trunks, and is the only special access, two-wire interface on which Direct Inward Dialing (DID) is available. The CPE&A may operate in wink start, delay-dial / start-dial, or immediate start mode, indicating the trunk supervisory signaling state by reversing the battery polarity (wink start is recommended). The network will, at the customer's option, transmit address information toward the CPE&A in the form of dial pulses (as described in Section 3.1.1) or DTMF digits (as in Section 3.2.1). Dial pulses will be conveyed by loop pulsing; battery and ground pulsing is not available. Call control procedures and relevant timing parameters are described in Section 2 and its subsections. CPE&A must provide any appropriate call progress tones (e.g., audible ring tone, busy, etc.), and must return an answer signal (i.e., battery reversal) within 500 ms. after the station party has answered.

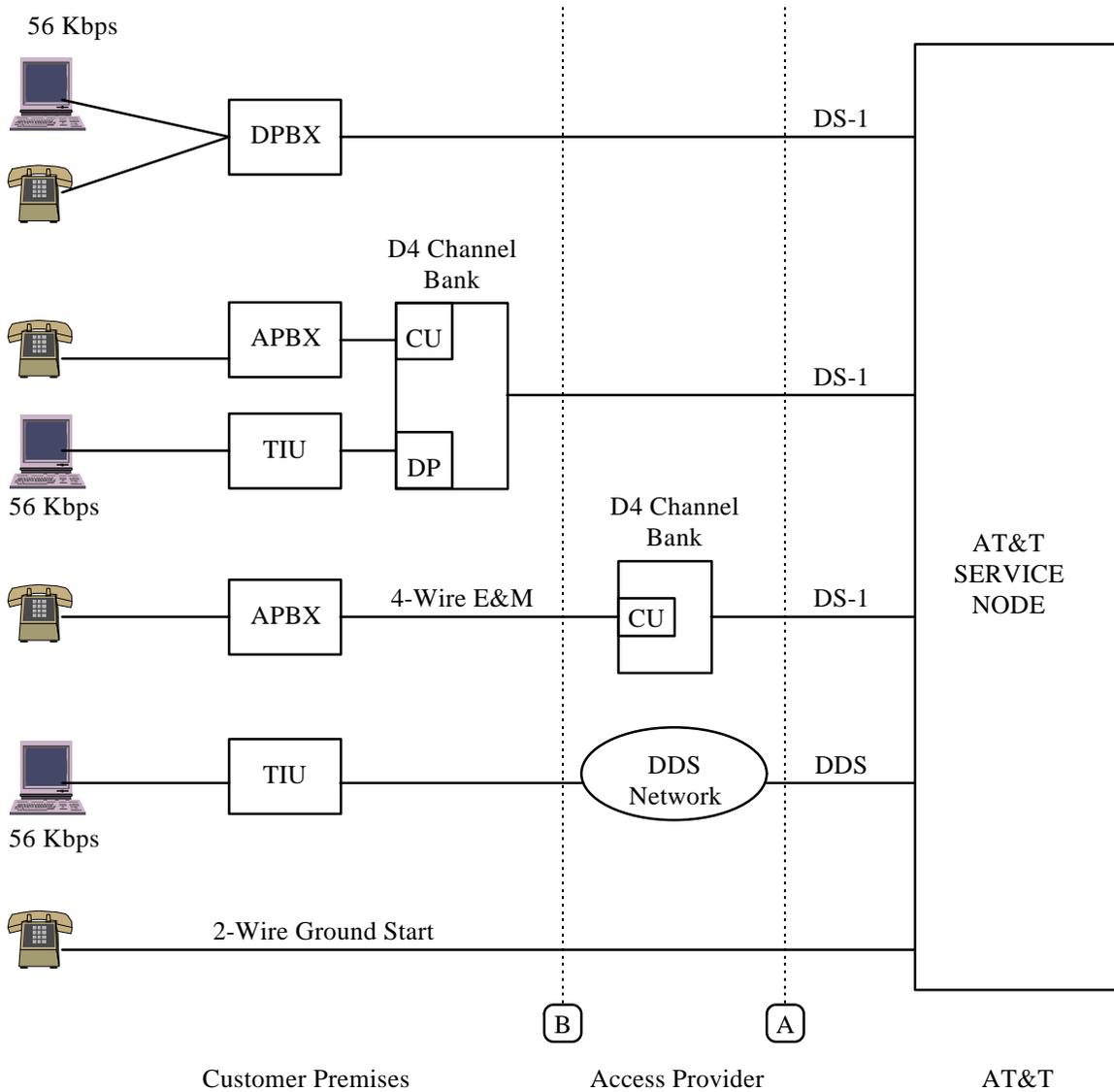


Figure II - 1. Several examples of special access interfaces to an ACP

(other access interfaces, as specified in Table II - 1 and Table II - 2 are also available)

1.2 Four-Wire E&M Trunks

Four-wire E&M trunks are recommended for all new special access installations requiring analog interface facilities. The material included here describes the termination characteristics presented to

the 4-Wire E&M facility by the network ACP, with regard to voiceband transmission and supervisory signaling. Call Control procedures relevant to E&M trunks may be found in Section 2 and its subsections.

1.2.1 Voiceband Transmission

The convention for which particular pairs the network ACP will transmit and receive on is described in Bell Communications Research Technical Reference TR-NPL-000335. The ACP will present nominal source and load impedances of 600 ohms, resistive, measured at the POT. Transmission levels for trunks are given in Section 5.

1.2.2 Supervisory Signaling

AT&T supports E&M signaling types I, II, and III, but some access providers may choose not to offer all three. Each of these signaling mechanisms is described in Bell Communications Research Technical Reference TR-NPL-000335.

1.3 Trunks Provided Over T1.5 Access

The DS-1 access arrangement allows connection of CPE directly to a Network Action Point switch at the carrier level. Figure II - 1 shows typical access architectures. Examples of CPE which are served by the DS-1 special access include (but are not limited to) Private Branch Exchanges (PBXs) and Automatic Call Distributors (ACDs). The PBX or ACD may connect to the network directly (if equipped with DS-1 trunk interfaces), or may connect to the network via a customer-provided digital channel bank (if the PBX or ACD is only equipped with analog trunk interfaces).

Currently, all DS-1 special access interfaces within the scope of this section employ either Superframe (SF, also known as D4), or Extended SuperFrame (ESF) framing format and SF channel sequence format. Bipolar with 8-Zero Substitution (B8ZS) line coding is preferred, but is not universally available on access links. If B8ZS is not available, Alternate Mark Inversion (AMI) line coding is used instead. Voice and voiceband data are encoded using mu-255 pulse code modulation (PCM).

AT&T will be the master timing source, providing synchronization traceable to the Primary Reference Source (PRS). The access facilities and the customer's CPE must therefore derive timing (i.e., loop-time) from AT&T's transmitted signal. For a discussion of loop timing, see AT&T Technical Reference TR 62411. Further information on digital network synchronization may be found in AT&T Technical Reference TR 60116^[6].

1.3.1 Information Encoding

This section contains specifications for encoding end-to-end user information (i.e., PCM voice or digital data). Supervisory signaling is covered in Section 1.3.2.

1.3.1.1 Pulse Code Modulation

AT&T Technical References TR 43801^[2] and TR 54016^[3] provide a complete description of the SF and ESF framing standards, channel format and PCM encode/decode methods. Also see Section 5 of this document, and ANSI standard EIA/TIA-464B for transmission levels and encoding/decoding loss guidelines.

1.3.1.2 56 Kbps Digital Data

In addition to voice calls, 56 Kbps, synchronous, circuit switched digital data calls are possible when the T1.5 access is used. This capability, which is used by the SDN and Switched Digital 56 Services, is full duplex unless the 56 Kbps digital broadcast capability is used (the 56 Kbps digital broadcast capability is described in Part I).

Each word (i.e., "octet") of a time slot consists of eight bits, numbered 1 through 8. Bit 8, which is the least significant bit, is periodically robbed for signaling use and is subject to over-writing when cross-connect equipment is employed in the access. User data may occupy bits 1 through 7. To assure network compatibility and end-to-end compatibility (which may involve dissimilar CPE), AT&T recommends that vendors adhere to the following conventions:

- a) Binary "1"s are represented by pulses and binary "0"s are represented by the absence of pulses.
- b) Once a data call is set up, a binary "1" is transmitted in the eighth bit of all words of the relevant time slot (including words appearing in both signaling and non-signaling frames).
- c) No all-zero words are transmitted in any time-slot. This constraint, known as the "per-channel pulse density constraint," is met while the call is in progress by observing recommendation (b) of this section. During call set-up, or while the trunk is idle, any method of per-channel pulse density enforcement is acceptable, provided it does not interfere with normal framing, formatting, and supervisory signaling.
- d) By convention, bits of information appearing in a time slot word are ordered chronologically, with bit 1 of the word representing the first bit of user data, and bit 7 of the word representing the last bit of user data.

1.3.2 Supervisory Signaling

Robbed bit supervisory signaling is used on all DS-1 special access interfaces within the scope of this document. Two robbed bit channels (A and B) exist within SF framing and formatting, and four (A, B, C, and D) exist within ESF. The ACP uses the A/B-bits (SF) or A/B/C/D bits (ESF) in a fashion that is consistent with E&M channel units. Thus, only two supervisory states are used: "on-hook" and "off-hook." These are summarized below.

1.3.2.1 On-Hook

- a) The CPE must transmit signaling bits A=B=0 (SF) or A=B=C=D=0 (ESF) as an on-hook indication.
- b) The CPE must interpret a "0" in the received signaling bits as an on-hook indication from the ACP.

1.3.2.2 Off-Hook

- a) The CPE must transmit signaling bits A=B=1 (SF) or A=B=C=D=1 (ESF) as an off-hook indication.
- b) The CPE must interpret a "1" in the received signaling bits as an off-hook indication from the ACP.

1.3.2.3 False Off Hook

A prolonged Off-hook, without subsequent dialed digits being received, may be interpreted by the ACP as a permanent seizure, and a fault on the channel, which may result in the channel being removed from service.

1.4 56 Kbps Single Channel

The 56 Kbps single channel special access is currently used for both the SDN and Switched Digital 56 Services. (See Part I for a brief description of these services.)

Figure II - 1 includes one example of a typical 56 Kbps single channel installation. Here, a signaling conversion function at the AT&T Point Of Presence (POP) maps the robbed bit signaling present on the DS-1 facility into the Data and Control modes supported by the DATAPHONE[®] Digital Service (DDS). This fact allows use of conventional, unmodified DDS facilities (without secondary channel) for access. The DDS access facilities extend from the AT&T Point of Presence to a Terminal Interface Unit (TIU) on the customer's premises. The TIU adapts the DDS line coding to a standard DCE/DTE interface (such as V.35 or RS-449/422), and performs the call control procedures described in Section 2. AT&T TR 62310 describes the DDS interface, including the Data and Control modes used to convey the supervisory signaling.

Call control procedures are consistent with Section 2 and its subsections, with some restrictions. These are explicitly noted where applicable. To encourage uniformity among access trunks, AT&T recommends the use of wink-start procedures in the outward (from CPE&A) direction. In the inward direction, immediate start is typical on 56 Kbps single channel non-DID trunks, and Delay-Dial / Start-dial (with special requirements on delay-dial signal duration as noted in Section 2.3.2) must be used on 56 Kbps single channel DID trunks.

1.4.1 Information Encoding

User data is encoded onto the DDS loop as described in AT&T TR 62310. That document describes how data is encoded in a bipolar, return-to-zero format, with a positive or negative pulse representing a binary "1" and the absence of a pulse representing a binary "0."

AT&T network equipment will interpret a Zero-code Suppression (ZS) sequence (see Figure II - 2b) as a string of seven consecutive binary "0"s in the user's data. If the CPE&A attempts to transmit more than six consecutive, unsuppressed "0"s, the user data may be corrupted. Zero-code Suppression sequences are also transmitted by the network toward the customer. It is possible that the network may transmit as many as twelve consecutive unsuppressed "0"s toward the CPE&A. Except for ZS sequences, no bipolar sequences are supported while a call is in progress. For example, Control Mode Idle (CMI) is not supported as an end-to-end signaling mechanism (as it is in conventional DDS), since the single channel access procedures use CMI for supervisory signaling, as described below. Only the 56 Kbps data rate is supported.

The equipment at the AT&T Point of Presence transmits and receives in synchronism with the Basic Synchronization Reference Frequency.

1.4.2 Supervisory Signaling

1.4.2.1 On-Hook

- a) The network will interpret Control Mode Idle (CMI) codes as indicative of the on-hook supervisory state. Figure II - 2a illustrates a typical CMI sequence.

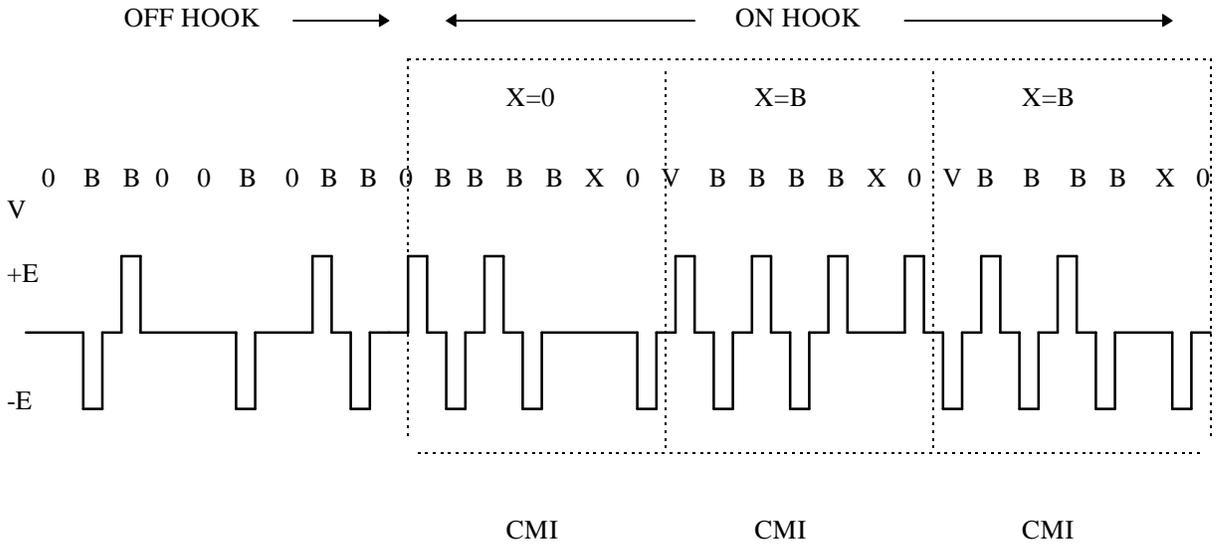
- b) The network will transmit CMI codes as an on-hook indication from the ACP.

1.4.2.2 Off Hook

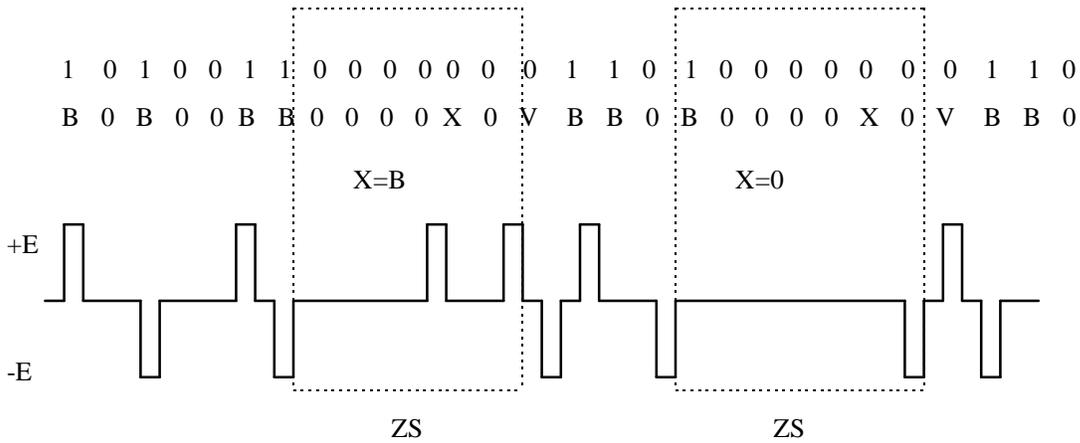
- a) The network will interpret the Data Mode as indicative of the off-hook supervisory state. A data mode signal is characterized by one of the following two attributes:
 - i. Absence from bipolar violations.
 - ii. Zero-code Suppression (ZS) sequences, as in Figure II - 2b.
- b) The network will transmit in Data Mode as an off-hook indication from the ACP.

1.4.3 Termination of DDS Access Facility

The customer has the responsibility of maintaining the proper TIU termination on the DDS access facility at all times. Removal of the termination, either by disconnecting or turning off power, causing the loss of CMI may be interpreted by the ACP as a fault on the channel, which may result in the channel being removed from service.



a) Control Mode Idle (CMI)



b) Zero Code Suppression (ZS)

- 0 Denotes zero volts transmitted - binary zero
- B Denotes $\pm E$ volts transmitted (polarity determined by the bipolar rule) - binary one.
- V Denotes $\pm E$ volts transmitted (polarity in violation of the bipolar rule) - binary one.
- X Equals 0 or B if the number of "B"s since the last "V" is odd or even, respectively. In example (a), the number of "B"s since the last "V" (not shown) until the first "X" (shown) is assumed to be odd. In example (b), it is assumed to be odd.

Figure II -2. Bipolar violation sequences on 56Kbps single channel access.

Table II - 1. Allowed combinations of address and method of operation on E&M

Directionality	Inward (to CPE&A)		Outward (from CPE&A)	
	Start	Address	Start	Address
1W OG			Wink	DP
1W OG			Wink	MF
1W OG			IS	DP
1W OG			Wink	DTMF
1W OG			Dial Tone	DP or DTMF (per call)
1W IC	Wink	DP		
1W IC	DDSD	DP		
1W IC	Wink	MF		
1W IC	DDSD	MF		
1W IC	IS/S-G	DP		
1W IC	IS	Supp.		
1W IC	Wink	Supp.		
1W IC	DDSD	Supp.		
1W IC	Wink	DTMF		
2W	Wink	DP	Wink	DP
2W	Wink	DP	Wink	MF
2W	Wink	MF	Wink	DP
2W	Wink	MF	Wink	MF
2W	DDSD	DP	Wink	DP
2W	DDSD	DP	Wink	MF
2W	DDSD	MF	Wink	DP
2W	DDSD	MF	Wink	MF
2W	Wink	Supp.	Wink	DP
2W	DDSD	Supp.	Wink	DP
2W	IS	Supp.	Wink	DP
2W	IS	Supp.	Wink	MF
2W	Wink	DTMF	Wink	DTMF
2W	IS/S-G	DP	Dial Tone	DP or DTMF (per call)
2W	DDSD	DP	Dial Tone	DP or DTMF (per call)
2W	Wink	DP	Dial Tone	DP or DTMF (per call)

Notes:

1. "Incoming" and "outgoing" refer to the CPE&A viewpoint.
2. CPE&A may be administered for DDSD instead of wink start in the outward direction, since the wink sent by the ACP will be interpreted by the CPE&A as a DDSD signal. The glare resolution procedures, however, are faster for wink start than for DDSD. It is therefore recommended that whenever possible, CPE&A be administered for wink start instead of DDSD in the outward direction.

Part II - Special Access Interconnection at the AT&T Point of Termination

Abbreviations (Directionality):

- 1W OG One way outgoing access line.
- 1W IC One way incoming access line.
- 2W Two way access line.
- DP Dial Pulse signaling.
- Supp. Suppressed outpulsing of address information.
- IS Immediate Start.
- IS/S-G Immediate Start / Stop-Go.
- DDSD Delay-Dial / Start-Dial.

Abbreviations (Address Signaling):

- MF Multifrequency signaling.
- DTMF Dual-tone multifrequency.

Abbreviations (Supervisory Signaling):

- E&M Ear and Mouth
- LS Loop Start
- GS Ground Start

Table II - 2. Allowed combinations of address and start-dial signaling on 56 Kbps single channel special access lines.

Directionality	Inward (to CPE&A)		Outward (from CPE&A)	
	Start	Address	Start	Address
1W OG			Wink	DP
1W IC	IS	Supp.		
1W IC	DDSD	DP		
2W	DDSD	DP	Wink	DP
2W	IS	Supp.	Wink	DP

Notes:

1. "Incoming" and "outgoing" refer to the CPE&A viewpoint.
2. CPE&A may be administered for DDS-D instead of wink start in the outward direction, since the wink sent by the ACP will be interpreted by the CPE&A as a DDS-D signal. The glare resolution procedures, however, are faster for wink start than for DDS-D. It is therefore recommended that whenever possible, CPE&A be administered for wink start instead of DDS-D in the outward direction.

Abbreviations:

IW OG	One way outgoing access line.
IW IC	One way incoming access line.
2W	Two way access line.
DP	Dial Pulse signaling.
Supp.	Suppressed outpulsing of address information.
IS	Immediate Start.
DDS-D	Delay-Dial Start-Dial.

2. CALL CONTROL PROCEDURES

This section includes call set-up scenarios relevant to special access. With caveats explicitly noted, and subject to availability as noted in Table II - 1 and Table II - 2, the procedures are applicable to two-wire loop reverse-battery, four-wire E&M, DS-1, and 56 Kbps single channel special access trunks.

It is possible to mix signaling types on a given trunk (e.g., DDS-D inward with wink start outward), but some signaling combinations are not allowed. Table II - 1 is a comprehensive list of signaling type combinations allowed on E&M trunks and DS-1 trunks. Combinations allowed on 56 Kbps single channel special access trunks are listed in Table II - 2.

Glare is the condition that exists when two switches each seize the same two-way trunk at about the same time. Glare detection and resolution are specific to each call control procedure, and are described in the relevant subsection. If a two-way trunk is shared by MEGACOM[®] and/or Toll Free MEGACOM[®], the ACP will resolve the glare. Otherwise, it is possible for a customer to choose whether the glare will be resolved by the CPE&A or by the ACP. If the CPE&A is able to detect and resolve glare, it is usually advantageous that it do so on trunks not shared by MEGACOM and/or Toll Free MEGACOM. Methods for detecting and resolving glare are given for wink start call control in Section 2.2.3 and for delay-dial / start-dial call control in Section 2.3.3.

For various causes, known or unknown, there may be short changes (called hits) in the received supervisory state. These may be either on-hook hits or off-hook hits. The ACP is reasonably

immune to hits. Typically, hits are of relatively short duration (compared with normal call control signals), and the ACP will ignore them on the basis of their short duration. Appropriate hit timing varies with circumstance, and is specified where applicable. In any case, the ACP will ignore spurious signal transitions of duration less than 1 ms.

All flash timing should be handled within the CPE&A. No flashes should be presented to the ACP.

In the outward (i.e., from CPE&A) direction, all trunks use Direct Outward Dialing (DOD). In the inward direction, trunks may use Direct Inward Dialing (DID) or outpulsing may be suppressed at the ACP.

2.1 Idle State

When the trunk is idle, both the CPE&A and the ACP are on-hook. The idle condition exists until either the CPE&A or the ACP seizes the trunk.

2.2 Wink Start

Wink start is the preferred method of call control because it inherently offers an integrity check on the trunk, and because it provides for rapid detection and resolution of glare on two-way trunks .

2.2.1 Trunk Seized by Calling CPE&A

From the idle state, the ACP will interpret a change in the CPE&A supervisory state to off-hook as a trunk seizure if the off-hook signal persists for at least 150 ms (30-40 ms if hit timing is specified). The ACP will continue to monitor the supervisory state of the trunk, and will interpret any on-hook transitions as one of the following indications, depending upon the stage of the call processing and the duration of the on-hook excursion:

- a) dial pulses (during the addressing phase of the call, if dial pulse signaling is provisioned).
- b) a disconnection (if the call completes normally).
- c) call abandonment before completion.
- d) glare resolution by the CPE&A (assuming the CPE&A is the glare-resolving party).
- e) a hit.

The ACP will send the leading edge of the wink within 150 ms. after detection of the seizure. The wink, as generated by the ACP, is a momentary, timed, off-hook signal, 140 to 290 ms. in duration.

The ACP will "blind" itself for the 70 ms. immediately following the trailing edge of the wink. After the 70 ms. elapses, the ACP will be ready to register address digits in the form of dial pulses (Section 3.1.2), dual tone multifrequency (Section 3.2.2), or multifrequency (Section 3.3.2) on E&M or DS-1 trunks. On 56 Kbps single channel access trunks, only dial pulse is allowed. The ACP must receive the first digit from the CPE&A within the first 5 seconds for MF and 10 seconds for DTMF, immediately following the trailing edge of the wink. For DTMF, 15 seconds are allowed between digits, for MF, all digits after the first group of 4 digits (KP + 3D), must be received within 15 seconds.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel. For single channel access however,

depending on the particular access architecture, this call progress information may be either blocked, or not PCM byte synchronized. Therefore, the CPE&A should not always expect to receive standard PCM encoded call progress information, nor expect to receive uninterrupted CMI.

If answer supervision to the calling CPE&A is provided, the ACP will transmit an off-hook signal toward the CPE&A to indicate that the far-end CPE&A has answered the call. The answer signal will last for at least 2 seconds.

Answer delay varies with application. On a voice call, the answer delay depends on the human response time. On a 56 Kbps data call, the called CPE&A often answers automatically, without need for human interaction. Vendors of CPE&A for the 56 Kbps digital data capability are encouraged to provide an answer-delay timer, the duration of which can be adjusted by the customer. For a 56 Kbps call to a domestic destination, an answer delay of about 30 seconds appears appropriate. International call set-up times are longer than domestic call set-up times and may be highly variable. Vendors of CPE&A for use with the international 56 Kbps data capability should consider providing an answer-delay timer which is adjustable to at least 90 seconds.

2.2.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook.

The ACP expects to receive a wink from the called CPE&A within 5 seconds. At this stage of the call, the ACP will regard a momentary off-hook signal as a wink if it lasts longer than 100 ms. but less than 350 ms. The ACP will ignore spurious signal transitions following the start or end of a wink for at least 3 ms. If no wink is received, the ACP will declare a wink time-out. Wink time-outs are logged at the ACP, since they indicate a problem with the facility or the far-end equipment (i.e., the CPE). Repeated wink time-outs can cause the trunk to be put in a "lockout" state.

On Direct Inward Dialing (DID) trunks, the ACP will begin to transmit address digits at least 70 ms. after it receives the wink. Address digits may be in the form of dial pulses (Section 3.1.1), dual tone multifrequency (Section 3.2.1), or multifrequency (Section 3.3.1).

When and if the destination party answers, the CPE&A must return answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers, but not sooner than 70 ms. after the trailing edge of the wink. The CPE&A should maintain this off-hook signal until the call is disconnected.

2.2.3 Glare Resolution

If the ACP seizes the trunk at about the same time as does the CPE&A, a glare condition exists. As explained in Section 2.2.1, the CPE&A expects the ACP to respond to a seizure by sending a wink. In a glare situation, however, the CPE&A detects an off-hook signal which persists too long to be a wink.

If the CPE&A is the glare-resolving party, then the ACP will continue to assert its trunk seizure; it will not return the trunk to the on-hook state, but will instead wait for the CPE&A to do so. The glare condition will persist until the CPE&A aborts its call attempt by returning its supervisory state to on-hook and preparing to receive address information from the ACP. The ACP will interpret the off-hook to on-hook transition as an indication that it may begin sending address digits.

If the ACP is the glare-resolving party, then it will return the trunk to the on-hook state after timing an off-hook signal from the CPE&A which lasts longer than 350 ms. 70 ms after the ACP returns the trunk to the on-hook state, it will be ready to register address digits.

2.3 Delay-Dial / Start-Dial

Delay-dial / start-dial is sometimes referred to as delay-dial with integrity check. Delay-dial / start-dial call control procedures are similar to those for wink start, except that different timing requirements apply. In particular, the CPE&A should observe different glare timing if it is the glare-resolving party on a two-way trunk. Moreover, the requirements on the duration of the delay-dial signal emitted by a CPE&A are different than the requirements on the duration of a wink.

2.3.1 Trunk Seized by Calling CPE&A

From the idle state, the ACP will interpret a change in the CPE&A's supervisory state to off-hook as a trunk seizure if the off-hook signal persists for at least 150 ms. The ACP will continue to monitor the supervisory state of the trunk, and will interpret any on-hook transitions as one of the following indications, depending upon the stage of call processing and the duration of the on-hook excursion:

- a) dial pulses (during the addressing phase of the call, if dial pulse signaling is provisioned).
- b) a disconnection (if the call completes normally).
- c) call abandonment before completion.
- d) glare resolution by the CPE&A (assuming the CPE&A is the glare-resolving party).
- e) a hit

The ACP will send the delay-dial signal (i.e., will change the supervisory state to off-hook) within 150 ms. after detection of the seizure. The delay-dial signal, as generated by the ACP, will normally last from 140 to 290 ms., but could last up to 4 seconds. The transition back to the on-hook state is known as the "start-dial" signal.

The ACP will "blind" itself for the 70 ms. immediately following the start-dial signal. After the 70 ms. elapses, the ACP will be ready to register address digits in the form of dial pulses (Section 3.1.2), dual tone multifrequency (Section 3.2.2), or multifrequency (Section 3.3.2) on E&M or DS-1 trunks. On 56 Kbps single channel access trunks, only dial pulse is allowed. The ACP must receive the first digit from the CPE&A within the first 5 seconds immediately following the start-dial signal. It must receive all the digits within the first 15 seconds immediately following the start-dial signal.

While the call is being set up, standard in-band call progress information (e.g. audible ring tone, busy, reorder, etc.) may be present on the transmission channel.

If answer supervision to the calling CPE&A is provided, the ACP will transmit an off-hook signal toward the CPE&A to indicate that the far-end CPE&A has answered the call. The network merely passes along all supervisory transitions as they occur.

Answer delay varies with application. On a voice call, the answer delay usually depends on the human response time. On a 56 Kbps data call, the called CPE&A often answers automatically, without need for human interaction. Vendors of CPE&A for the 56 Kbps digital data capability are encouraged to provide an answer-delay timer, the duration of which can be adjusted by the customer. For a 56 Kbps call to a domestic destination, an answer delay of about 30 seconds appears appropriate. International call set-up times are longer than domestic call set-up times and may be highly variable. Vendors of CPE&A for use with the international 56 Kbps data capability should consider providing answer-delay timer which is adjustable to at least 90 seconds .

2.3.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook.

The ACP expects to receive a delay-dial signal from the called CPE&A within 5 seconds. On four-wire E&M trunks and on DS-1 trunks, the ACP will regard an off-hook signal as a delay-dial signal if it lasts longer than 100 ms., but less than 4 seconds. On 56 Kbps single-channel access trunks, the ACP will regard an off-hook signal as a delay-dial signal if it lasts longer than 1 second, but less than 4 seconds. The ACP will ignore spurious signal transitions following the delay-dial and start-dial signals for at least 3 msec. If no delay-dial signal is received within 5 seconds, the ACP will declare a delay-dial time-out. Delay-dial time-outs are logged at the ACP, since they indicate a problem with the facility or the far-end equipment (i.e., the CPE). Repeated delay-dial time-outs can cause the channel to be removed from service.

On Direct Inward Dialing (DID) trunks, the ACP will begin to transmit address digits at least 70 ms. after it receives the delay-dial signal. Address digits may be in the form of dial pulses (Section 3.1.1) or multifrequency (Section 3.2.1).

When and if the destination party answers, the CPE&A must return answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers. On trunks with suppressed outpulsing, it should not be returned sooner than 70 ms. after the start-dial signal. The CPE&A should maintain this off-hook signal until the call is disconnected.

2.3.3 Glare Resolution

If the ACP seizes the trunk at about the same time as does the CPE&A, a glare condition exists. As explained in Section 2.3.1, the CPE&A expects the ACP to respond to a seizure by sending a delay-dial signal, followed by a start-dial signal. In a glare situation however, the CPE&A detects an off-hook signal which persists too long to be a delay-dial signal.

If the CPE&A is the glare-resolving party, then the ACP will continue to assert its trunk seizure; it will not return the trunk to the on-hook state, but will instead wait for the CPE&A to do so. The glare condition will persist until the CPE&A aborts its call attempt by returning its supervisory state to on-hook and preparing to receive address information from the ACP. The ACP will interpret the off-hook to on-hook transition as an indication that it may begin sending address digits.

If the ACP is the glare-resolving party, then it will return the trunk to the on-hook state after timing an off-hook signal from the CPE&A which lasts longer than 4 seconds. 70 ms. after the ACP returns the trunk to the on-hook state, it will be ready to register address digits.

2.4 Immediate Start

Because immediate start operation is not amenable to glare detection, it is never recommended that two-way trunks be provisioned for immediate start in both the inward and outward directions. Immediate start can be used in the inward direction on a two-way trunk with inward dialing suppressed, but the CPE&A should be the glare-resolving party¹. Immediate start operation provides no trunk integrity checking during senderized call set-up, and therefore its use on senderized one-way trunks is recommended only if the CPE&A cannot accommodate wink start or delay-dial / start-

1. This type of operation is common on 56 Kbps trunks; it is unusual on voiceband and voiceband data trunks.

dial operation. In the inward direction, stop-go operation is permissible with dial pulse signaling (see Section 2.4.2).

2.4.1 Trunk Seized by Calling CPE&A

From the idle state, the ACP will interpret a change in the CPE&A's supervisory state to off-hook as a trunk seizure if the off-hook signal persists for at least 150 ms. The ACP will continue to monitor the supervisory state of the trunk, and will interpret any on-hook transitions as one of the following indications, depending upon the stage of call processing and the duration of the on-hook excursion:

- a) dial pulses (during the addressing phase of the call, if dial pulse signaling is provisioned).
- b) a disconnection (if the call completes normally).
- c) call abandonment before completion.
- d) a hit.

The ACP maintains the on-hook supervisory state during this stage of call processing.

Within 65 ms. of detecting the seizure, the ACP will be ready to register address digits in the form of dial pulses (Section 3.1.2) or dual tone multifrequency (Section 3.2.2) on E&M or DS-1 trunks. The ACP must receive the first digit from the CPE&A within the first 15 seconds immediately following detection of seizure. It will allow a maximum of 15 seconds between each digit.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel.

If answer supervision to the calling CPE&A is provided, the ACP will transmit an off-hook signal toward the CPE&A to indicate that the far-end CPE&A has answered the call.

2.4.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook.

On Direct Inward Dialing (DID) trunks, the ACP will begin to transmit address digits at least 65 ms. after seizing the trunk. Address digits will be in the form of dial pulses (Section 3.1.1).

If stop-go operation is used, the ACP will suspend outpulsing of address digits upon detection of an off-hook signal (i.e., a "stop" signal) from the CPE&A. When the CPE&A's supervisory state returns to on-hook (i.e., a "go" signal), the ACP will resume outpulsing. Only one stop-go sequence is allowed per call origination and must not occur after the 4th and later digits.

When and if the destination party answers, the CPE&A must return answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers. On trunks with suppressed outpulsing, it should not be returned sooner than 200 ms. after the ACP seizes the trunk (if the answer signal is returned more promptly than 200 ms., the ACP may incorrectly regard it as a glare indication). The CPE&A should maintain this off-hook signal until the call is disconnected.

2.5 Dial Tone Start

Dial tone can be provided by the ACP to support non-senderized (cut-through) operation in the outward (from CPE&A) direction. On dial tone trunks, the ACP will accept either dial pulse or Dual Tone Multifrequency address signals, on a per-call basis. The ACP does not support dial tone start call control in the inward (to CPE&A) direction.

2.5.1 Trunk Seized by Calling CPE&A

From the idle state, the ACP will interpret a change in the CPE&A's supervisory state to off-hook as a trunk seizure if the off-hook signal persists for at least 150 ms (30-40 ms. if hit timing is specified). The ACP will continue to monitor the supervisory state of the trunk, and will interpret any on-hook transitions as one of the following indications, depending upon the stage of call processing and the duration of the on-hook excursion:

- a) dial pulses (during the addressing phase of the call, if dial pulse signaling is provisioned).
- b) a disconnection (if the call completes normally).
- c) call abandonment before completion.
- d) a hit.

The ACP will transmit dial tone to the CPE&A upon recognition of the seizure and will maintain the on-hook supervisory state during this stage of call processing. Address digits may be in the form of dial pulses (Section 3.1.2) or dual tone multifrequency (Section 3.2.2). The ACP must receive the first digit from the CPE&A within the first 15 seconds immediately following application of dial tone. It will allow a maximum of 15 seconds between each digit. Within 500 ms. after receiving the first digit, the ACP will remove the dial tone.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel.

If answer supervision to the calling CPE&A is provided, the ACP will transmit an off-hook signal toward the CPE&A to indicate that the far-end CPE&A has answered the call. The network merely passes along all supervisory transitions as they occur.

Note: When certain older switching offices are involved at the egress end of calls to non-special access destinations, a momentary off-hook signal may be returned as the call is being established. This off-hook signal will reach the calling CPE&A after the address digits have been sent, while the CPE&A is awaiting answer. If the CPE&A responds too quickly, this signal may appear as an answer signal followed by a disconnect. If such symptoms are experienced, the CPE&A should be adjusted so an answer signal is not considered valid until it persists for 4 to 5 seconds.

2.6 Disconnection

Disconnect procedures for calling CPE&A differ slightly from those for called CPE&A. This section lists the procedures for each, with these procedures further categorized according to which CPE&A has initiated the disconnect. Normally, a disconnect is initiated by either the calling CPE&A or the called CPE&A, but the reader is cautioned that international calls lasting longer than 9 hours might be torn down by the switching equipment in some foreign countries. If this happens, the effect will appear as if the distant CPE&A has initiated the disconnect.

2.6.1 Calling CPE&A Procedures

2.6.1.1 Calling CPE&A Hangs Up First

After the call is established, the ACP will regard an on-hook signal of at least 150 ms. duration from the calling CPE&A as a disconnect signal.

If answer supervision is provided, the ACP will go on hook toward the CPE&A within 1050 ms. after the ACP recognizes the disconnect signal. The ACP's on-hook signal is an indication that the call has been disconnected and the trunk is idle. The CPE&A should not attempt to originate another call on this trunk until the received on-hook signal has persisted for a guard time of at least 650 ms. The ACP may seize the trunk to process a new call as soon as 800 ms. after initiation of its on-hook signal.

Note: The network could seize the trunk immediately after an on hook condition. This should be no problem unless CPE design does not tolerate incoming seizures during guard interval.

2.6.1.2 Called CPE&A Hangs Up First

If the called CPE&A hangs up first, the call will not be disconnected (and the billing clock will not be stopped) until either:

- a) the calling CPE&A disconnects as described above, or
- b) a time-out period of up to 20 seconds elapses.

If the called CPE&A returns the trunk to an off-hook state before either of these events occurs, the call will not be disconnected. If answer supervision is provided to the calling CPE&A, the ACP will change its state to on-hook toward the calling CPE&A to indicate that the called CPE&A has hung up.

2.6.2 Called CPE&A Disconnect Procedures

2.6.2.1 Calling CPE&A Hangs Up First

If the calling CPE&A has initiated the disconnect (as described in Section 2.6.1), the ACP will change the supervisory state of the egress trunk to on-hook. When the called CPE&A receives an on-hook signal of at least 300 ms. duration, it should interpret a disconnect, and should return the trunk to the on-hook state within 250 ms. A slower response may be interpreted by the ACP as a new seizure.

2.6.2.2 Called CPE&A Hangs Up First

If the called CPE&A hangs up first, the call will not be disconnected until either:

- a) the calling CPE&A disconnects as described in Section 2.6.1, or
- b) a time-out period of up to 20 seconds elapses.

If the called CPE&A returns the trunk to an off-hook state before either of these events occurs, the call will not be disconnected. When the call is disconnected, the ACP will confirm this by changing the supervisory state of the trunk to on-hook for at least 300 ms.

3. ADDRESS DIGIT TRANSMISSION

Special access CPE&A may use Dual Tone Multifrequency (DTMF), Multifrequency (MF) or Dial Pulse (DP) addressing. To some extent, the addressing types may be intermixed on two-way trunks, but not all combinations are allowed. Moreover, some restrictions apply based on facility type and call control procedure type. Vendors and users should refer to Table II - 1 and Table II - 2 for a comprehensive list of allowed combinations.

3.1 Dial Pulse

Dial pulse address signaling consists of a sequence of momentary on-hook excursions (breaks), the number of such breaks corresponding to the value of a dialed digit (the exception is the digit "0", which is represented by a sequence of ten breaks). The dial pulse break interval is the time interval corresponding to the on-hook state of the trunk during dialing. The dial pulse make interval is the time interval corresponding to the off-hook state of the trunk during dialing. A break interval and an adjacent make interval constitute a dial pulse cycle. The percent break is the proportion of the dial pulse cycle occupied by the break interval.

Speed is expressed in pulses per second (pps), and it refers to the pulse repetition rate averaged over the transmission of a single digit, not including interdigital time. Equivalently, it is the reciprocal of the dial pulse cycle time.

Interdigit time is measured from the trailing edge (final break-to-make transition) of the last pulse of one digit to the leading edge (initial make-to-break transition) of the first pulse of the following digit. During the interdigital time, the trunk is off hook.

3.1.1 ACP Dial Pulse Generation

- a) The ACP will generate dial pulses with a percent break in the range of 58 to 64 percent (the break interval generated by the ACP to represent the digit "1" will be between 56 and 64 ms.).
- b) In generating a dial pulse digit, the ACP will pulse at a speed between 9.8 pps and 10.2 pps.
- c) The ACP will outpulse the first digit at least 70 ms after:
 - i) the detection of the start dial signal (on delay-dial / start-dial trunks).
 - ii) the detection of the trailing edge of the wink (on wink start trunks).
 - iii) initial seizure of the trunk by the ACP (on immediate start trunks).
- d) The ACP will send dial pulses with an interdigital time between 600-700 ms.
- e) The total duration of short makes and breaks at the initiation of a make interval will not exceed 3 ms.
- f) Any spurious break other than those in item e) above, during any off-hook interval while the ACP is in the address signaling state, will not exceed 1 ms. in duration (by definition, the address signaling state is entered upon detection of a start-dial signal or the trailing edge of a wink on delay-dial / start-dial trunks or wink start trunks, respectively. On Immediate start trunks, the address signaling state is entered upon seizure of the trunk).

- g) Spurious makes in the break interval of a dial pulse will not occur.

3.1.2 ACP Dial Pulse Reception

- a) The ACP will recognize dial pulses within the speed range of 7.5 pps to 12 pps.
- b) The ACP will properly interpret dial pulses having a percent break between 40 and 80 percent.
- c) The ACP will accept interdigital times as short as 300 ms.
- d) The ACP will regard short makes and breaks within 3 ms. after the initiation of a make interval as hits.
- e) During the address reception state, the ACP will regard spurious break intervals of less than 10 ms. as hits (by definition, the address reception state is entered upon dispatch of a start-dial signal or a wink on delay-dial / start-dial trunks or wink start trunks, respectively. On Immediate start trunks, the address reception state is entered upon detection of trunk seizure. The address reception state ends 600 ms. after the final digit is received).
- f) During the address reception state, the ACP will regard spurious make intervals of less than 1 ms. as hits.

3.2 Dual Tone Multifrequency

Dual tone Multifrequency (DTMF) address signaling is a method of signaling using the voice transmission path. In general, this method may use up to 16 distinct signals. Each signal is synthesized by superimposing two sinusoidal tones, one each from two geometrically-spaced groups of frequencies, designated the "low group" and the "high group." The ACP allows the use of 12 of these 16 signals to represent the digits 0 through 9, plus the octothorp ("#") and asterisk ("*") characters. Use of the alpha characters (A-D) is not allowed. Table II - 3 summarizes the frequency pairs used to represent the allowed DTMF digits.

Table II - 3. Component frequencies for DTMF digits.

DTMF CODES				
		HIGH GROUP FREQUENCIES (Hz)		
		1209	1336	1477
LOW GROUP	697	1	2	3
FREQUENCIES	770	4	5	6
(Hz)	852	7	8	9
	941	*	0	#

3.2.1 DTMF Digit Generation

Power per DTMF frequency component, as generated at the ACP and measured at the zero reference point, will be $-7 \text{ dBm}0 \pm 1 \text{ dB}$. The reader should adjust this value according to Table II - 6 to determine the tone levels present at the AT&T Point of Interface. On four-wire E&M trunks, with these constraints, the DTMF generators at the ACP meet or exceed the specifications provided in LSSGR and EIA/TIA-464B.

3.2.2 DTMF Digit Registration

DTMF receivers at the AT&T point of presence meet or exceed the specifications provided in LSSGR and those provided in EIA/TIA-464B for Type I receivers.

3.3 Multifrequency

The multifrequency (MF) signaling arrangements make use of pairs of frequencies out of a group of six frequencies. These frequencies are 700, 900, 1100, 1300, 1500, and 1700 Hz. On special access trunks, MF signals may be used for called number address signaling and calling number identification. Only the numerical signals (0 - 9) and the control signals KP and ST are allowed on special access connections. The allowed MF codes are summarized in Table II - 4.

Table II - 4. Multifrequency codes used on special access

COMPONENT FREQUENCIES (Hz)	DIGIT AND CONTROL SIGNALS
700, 900	1
700, 1100	2
700, 1300	4
700, 1500	7
900, 1100	3
900, 1300	5
900, 1500	8
1100, 1300	6
1100, 1500	9
1100, 1700	KP
1300, 1500	0
1500, 1700	ST

Multifrequency signaling is used with wink start or delay-dial / start-dial call control. Two-way trunks may not mix MF signaling in one direction with DTMF signaling in the other.

3.3.1 MF Digit Generation

The ACP, upon detecting the wink or start-dial signal, will transmit the required MF signals. The start of outpulsing will be delayed as little as possible, but will not be less than 50 ms. under normal conditions.

The MF transmitter at the ACP and its signal will meet the following requirements:

- a) The two frequencies of each code will start and end within 1 ms. of each other.
- b) Power per MF frequency component, as generated at the ACP and measured at the zero reference point, will be $-7 \text{ dBm} \pm 1 \text{ dB}$. The reader should adjust this value according to Table II - 6 to determine the tone levels present at the AT&T Point of Interface.
- c) There will not be over 1.0 dB difference between the powers of the two frequencies of any code.
- d) The total power of extraneous signal components will be at least 30 dB below the power level of either frequency of the code.
- e) The KP signal length will be 90 to 120 ms.
- f) The ST and digit signal lengths will each be 58 to 75 ms. (may be as short as 50 ms. from the network if 10 digits per second has been specified).
- g) The interval between signals will be 58 to 75 ms. (may be as short as 50 ms. from the network if 10 digits per second has been specified).
- h) The frequencies will be within ± 1.5 percent of stated nominal values.
- i) On four-wire E&M trunks, the transmitter will present a nominal source impedance of 600 ohms, resistive.
- j) On four-wire E&M trunks, the transmitter will have a longitudinal balance to ground at least equal to that required for voice transmission.
- k) During tone-off periods, transmitted power at any MF frequency will not exceed -58 dBm at the AT&T Point of Interface.

3.3.2 MF Digit Registration

MF receivers at the ACP will meet the following requirements:

- a) The receiver impedance will match that of the interface on which it is used (600 ohms, nonreactive on 4-wire analog trunks). Its single frequency return loss will be at least equal to that required for voice transmission.
- b) The receiver will respond to signal levels between 0 and -25 dBm per frequency measured at the ACP's zero reference point. Existing receivers may have a sensitivity of only -22 dBm , but new circuits will meet the -25 dBm requirement. The receiver will not respond if the signal level drops below -35 dBm per frequency.

- c) The receiver will not respond to address signals prior to being "unlocked" by receipt of a KP signal. If two or more consecutive KP signals are received, all those after the first one will be ignored.
- d) Once unlocked, the receiver will remain unlocked until it receives the ST signal.
- e) The receiver will respond to signals in which each frequency component duration is at least 35 ms. The receiver will respond to a KP signal that is at least 55 ms long. The two frequency components may be shifted in time relative to each other by as much as 4 ms.
- f) The receiver will not respond to signals shorter than the requirements in the preceding paragraph, and will not respond to signals in which the two components are not coincident for more than 10 ms.
- g) The receiver will recognize interpulse intervals as short as 25 ms. This interval is defined as the time during which no signal frequency component is above -35 dBm.
- h) After the minimum length signal has been received, the receiver will bridge interruptions as long as possible, consistent with meeting the interpulse requirement.
- i) The receiver will accept up to 10 digits per second. However, 7.5 digits per second is the recommended MF signaling rate, since this is the design center rate for most existing MF systems .
- j) The receiver will check for the presence of two, and only two, valid frequency components in each pulse. If a pulse fails to meet this requirement, the ACP will apply reorder treatment to the call.
- k) The receiver will tolerate pulses in which the power levels of the two frequency components differ by as much as 6 dB.
- l) MF receivers at the ACP will meet or exceed the same level of noise tolerance (message circuit noise and impulse noise) as do DTMF receivers. These requirements, and the method for measuring compliance with them, may be found in ANSI standard EIA/TIA-464B. In addition, MF receivers will tolerate the following types and levels of power line induction with an error rate of not more than one in 25,000 frequency-pair signals:

60 Hz	81 dBmC0
180 Hz	68 dBmC0
- m) The receiver will tolerate 2A-B and 2B-A modulation products caused by intermodulation when MF pulses are transmitted over standard carrier facilities. The power sum of these modulation products is expected to be at least 28 dB below each frequency component level of the signals.
- n) The longitudinal balance to ground will be equal to that required for voice transmission.
- o) The receiver will accept MF signals in which the component tones deviate in either direction by 5 Hz more than 1.5 percent of their nominal frequencies.

4. CALL PROGRESS SIGNALS

Table II - 5 summarizes call progress tones generated at the ACP for special access interfaces, and typical power levels at the ACP's zero reference point. The reader should adjust these values according to Table II - 5 to determine the tone levels present at the AT&T Point of Interface. These tones adhere to the telephone industry precise tone plan. The table applies only to tones which are generated by the ACP. Call progress tones generated within the CPE (e.g., busy tone) are not processed by AT&T network equipment, but are instead passed through the network transparently. Power levels at the CPE interface may differ from those at the AT&T POT because of loss in the access facility.

Table II - 5. Cadences, frequency components, and power levels (per frequency) of call progress signals generated at the ACP.

Tone	Cadence	Level (dBm ₀ ± 1.5 dB)				
		350 Hz	440 Hz	480 Hz	620 Hz	Combined
Dial Tone	Steady On	-14	-14			-11
Busy Tone	0.5 sec. on, 0.5 sec. off			-19	-19	-16
Reorder Tone	0.25 sec. on, 0.25 sec. off			-19	-19	-16

5. TRANSMISSION LOSS AND LEVELS

This section contains information relative to transmission levels at the AT&T point of termination. It is the responsibility of CPE vendors to assess the impact of this information on equipment design requirements. Likewise, it is the responsibility of the customer to engineer access facility requirements except when total service or coordinated access is requested from AT&T.

5.1 Other Technical References and Standards

End-to-end transmission characteristics involve a number of non-AT&T components. These may include CPE and access facilities, each of which are outside the scope of this document. ANSI standards recommend CPE design criteria which are generally applicable to special access as well as switched access. Bell Communications Research references provide information on access facilities which are typically available from a Local Exchange Carrier (LEC). Finally, AT&T technical references for the ISDN Primary Rate Interface look to this document for information regarding transmission levels. Each of these is discussed briefly in the following subsections.

5.1.1 ANSI Standard EIA/TIA-464B

EIA/TIA-464B is a technical specification of pre-ISDN PBX requirements for voiceband applications. The information contained in it regarding PBX loss and level requirements may

nevertheless be applied to a wider class of CPE, including that used for special access. It is the recommended reference on special access CPE insertion loss and transmission levels.

5.1.2 Bell Communications Research TR-NPL-00035, TR-NPL-00034 and PUB 62508

Technical References TR-NPL-00035 and TR-NPL-00034 provide information on analog, special or switched access facilities generally available from Local Exchange Carriers. These documents catalog the available loss ranges, interface configurations, conditioning, and other relevant data. Analog interface codes provided in Table II - 6 are explained in TR-NPL-00035.

PUB 62508 provides information on Digital special access facilities generally available from Local Exchange Carriers. The interface code provided in Table II - 6 for the DS-1 interface is explained in PUB 62508.

5.1.3 AT&T TR 41459

TR 41459 describes criteria for CPE compatibility with the AT&T Primary Rate Interface (PRI). This document focuses mainly on requirements unique to ISDN, and discusses neither transmission levels nor other parameters common to 8-bit PCM transmission characteristics. However, Domestic ISDN B-channels with a speech bearer capability employ 8-bit, mu-255 PCM encoding, and adhere to the same network loss plan as do any other digital special access trunks. The mu-law transmission guidelines of this section (and of EIA/TIA-464B) are therefore applicable to speech-bearing PRI B-channels. To achieve satisfactory end-to-end transmission quality, transmission levels and inserted connection loss (ICL) must be carefully controlled. Customers who do not request total service or coordinated access are responsible for engineering their own access. Among other things, this includes specifying the desired inserted connection loss.

5.1.4 Relationship Between Analog and PCM Transmission Levels

The relationship of digital (i.e., PCM) encoder levels to analog power levels is conventionally defined using a deterministic sequence of PCM words known as the "digital milliwatt." This sequence has been accepted by the ITU-T (Rec. G.711), and is tabulated in Figure II - 3. If driven by a repeated pattern of digital milliwatt at its PCM input, a zero level mu-255 decoder is defined such that it would emit a 0 dBm, 1000-Hz sinusoid. However, note that the digital milliwatt may not necessarily result from driving a zero-level mu-255 encoder, since phase shifts and slight frequency deviations would probably give rise to another sequence of PCM words. A more useful signal, the Digital Reference Signal (DRS), is any sequence of PCM words which would be decoded by a zero level mu-255 decoder into a 0 dBm sinusoid having a frequency between 1000 Hz and 1020 Hz (usually 1004 Hz). A zero level mu-255 encoder would yield a DRS when driven by a 0 dBm sinusoid with a nominal frequency of 1004 Hz. In other words, the DRS has the equivalent analog power content as the digital milliwatt, but its frequency is loosely specified and its phase shift is not specified at all.

	BIT NUMBER							
	1	2	3	4	5	6	7	8
1	0	0	0	1	1	1	1	0
2	0	0	0	0	1	0	1	1
3	0	0	0	0	1	0	1	1
PCM	4	0	0	0	1	1	1	0
Words	5	1	0	0	1	1	1	0
	6	1	0	0	0	1	0	1
	7	1	0	0	0	1	0	1
	8	1	0	0	1	1	1	0

Figure II -3. The Digital Milliwatt.

Note: The digital milliwatt is useful as a basis for definition, and this document uses it in that context. However, the digital milliwatt should not be used as a maintenance test tone. The recommended maintenance test tone is the DRS defined above. The DRS is generated by several commercially-available test sets, and may be used for aligning carrier systems and for other circuit installation and maintenance procedures.

5.1.5 Transmission Levels at the AT&T Point of Termination

Table II - 6 gives the transmission levels at the AT&T point of termination service node for the two-wire ground start, and four-wire E&M, network interfaces. In the "AT&T Receive" direction, the Table gives the analog test signal from the CPE&A, present at the point of termination service node, that would cause a digital reference signal to be present at the ACP's zero reference point. In the "AT&T Transmit" direction, the Table gives the analog signal that would be presented to the CPE&A at the point of termination service node, if a digital reference signal was applied at the ACP's zero reference point. Table II - 6 also states that if a digital reference signal is present in a DS-1 facility at the point of termination, then a digital reference signal will also be present at the ACP's zero reference point (and the converse). The interface codes in Table II - 6 are defined in Bell Communications Research publications TR-NPL-000335 and PUB 62508.

The 56 Kbps single-channel access is intended for use with purely digital, lossless services, so transmission levels are not applicable.

Table II - 6. Transmission Levels at the Point of Termination Service Node

Special Access Transmission Levels²			
Interface	Code	AT&T Transmit at POT SN³	AT&T Receive at POT SN⁴
2LS, 600 ohm	02LO2	-2 dBm	-2 dBm
2LS, 900 ohm	02LO3	-2 dBm	-2 dBm
2GS, 600 ohm	02GO2	-2 dBm	-2 dBm
2GS, 900 ohm	02GO3	-2 dBm	-2 dBm
2LRB, 600 ohm	02RV2-T	-2 dBm	-2 dBm
4E&M, Type I	06EA2-M	-16 dBm	+7 dBm
4E&M, Type II	08EB2-M	-16 dBm	+7 dBm
4E&M, Type III	08EC2-M	-16 dBm	+7 dBm
DS-1	4DS9-15	DRS	DRS
56 Kbps Digital	6DU5-56	N.A.	N.A.

2. Analog values are approximate power levels of 1004-Hz sinusoids.

3. These analog output levels result if digital reference signal is applied at ACP's zero reference point.

4. DRS results at ACP's zero reference point when 1004-Hz sinusoids with these levels are presented to the AT&T POT.

REFERENCES

-
- 1 AT&T, Digital Synchronization Network Plan, Technical Reference 60110.
 - 2 AT&T, Digital Channel Bank Requirements and Objectives, Technical Reference 43801.
 - 3 AT&T, Extended Superframe Format Description and Interface Specifications, Technical Reference 54016.

Special Access Connections

To The

AT&T Network

Part III

CPE Guidelines

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1. INTRODUCTION

This part of TR 41458 does not provide interface specifications to AT&T's network. It has been included as a part of this document solely as an aid for CPE designers. Material in Part III is based on assumptions about access facilities that AT&T would typically engineer under total service or coordinated access arrangements. Some sections of Part III may not be applicable or may not be appropriate if other special access arrangements, such as customer-provided access are used.

Any special access testing that AT&T performs in support of CPE vendors who are building equipment for connection to AT&T's network will be based on the guidelines contained in this Part. CPE vendors should recognize that the words "should" and "must," as used in Part III, are only intended to suggest AT&T's expectations of CPE under standard engineering rules that apply to equipment connecting to the AT&T network.

2. SPECIAL ACCESS FACILITIES

This section includes recommendations on design criteria for CPE that connects to an ACP via special access. This information is intended to be consistent with ANSI standard EIA/TIA-464B, but augments that material in several areas. These areas include the four-wire E&M Type III interface and the 56 Kbps single-channel digital interface. Material in this section is based on certain assumptions about the access facility characteristics, and may not be valid for CPE used on facilities not adhering to those assumptions.

The material is organized according to the facility type (2-wire loop start, 2-wire ground start, 2-wire loop reverse-battery, 4-wire E&M, DS-1, and 56 Kbps single channel special access.) Although the signaling mechanisms for the latter four trunk types are different, the call control procedures that use these signaling mechanisms are, with few exceptions, identical. Therefore, call control procedures are not covered in this section, but may be found in Section 3.

2.1 Two-Wire Analog Interfaces

2.1.1 Ground Start and Loop Start

Ground start and loop start trunks are adequately treated in EIA/TIA-464B, and are not discussed further in this document.

2.1.2 Loop Reverse-Battery

The discussion of loop reverse-battery operation given in EIA/TIA-464-B relative to DID trunks applies, but vendors should also note the clarifications given in Part II of this document.

2.2 Four-Wire E&M Trunks

E&M signaling conveys trunk supervisory signaling between the ACP and the CPE with signaling wires that are separate from, but associated with the transmission loop. Thus, a four-wire E&M trunk consists of two voice transmission pairs (labeled T, R and T₁, R₁) plus the E signaling lead, the M signaling lead, and (sometimes) leads for signal battery (SB) and signal ground (SG).

2.2.1 Voice and Transmission

- a) The CPE should transmit over the T, R pair.
- b) The CPE transmitter should present a source impedance of 600 ohms, resistive to the network interface.

- c) The CPE should receive on the T, R₁ pair.
- d) The CPE receiver should present a load impedance of 600 ohms, resistive to the network interface.
- e) The CPE should conform to the loss and level plan specified in EIA/TIA-464B.
- f) The called CPE should provide all call progress tones, except for appropriate network reorder tones, network busy tones, and network recorded announcements.
- g) Power ringing current is not provided over E&M trunks, and should be generated locally by the CPE.

2.2.2 E&M Type I Supervisory Signaling

The Type I interface is adequately treated in ANSI specification EIA/TIA-464B, and is not discussed further in this document.

2.2.3 E&M Type II Supervisory Signaling

The Type II interface is adequately treated in ANSI specification EIA/TIA-464B, and is not discussed further in this document.

2.2.4 E&M Type III Supervisory Signaling

The Type III interface (Figure III - 1) is a compromise, partially looped, 4-wire E&M lead arrangement. It is almost identical to the Type I interface except that the battery and ground for signaling on the M lead are supplied by the network over the SB and SG leads, respectively. Type I and Type III E-lead circuits are identical.

2.2.4.1 General

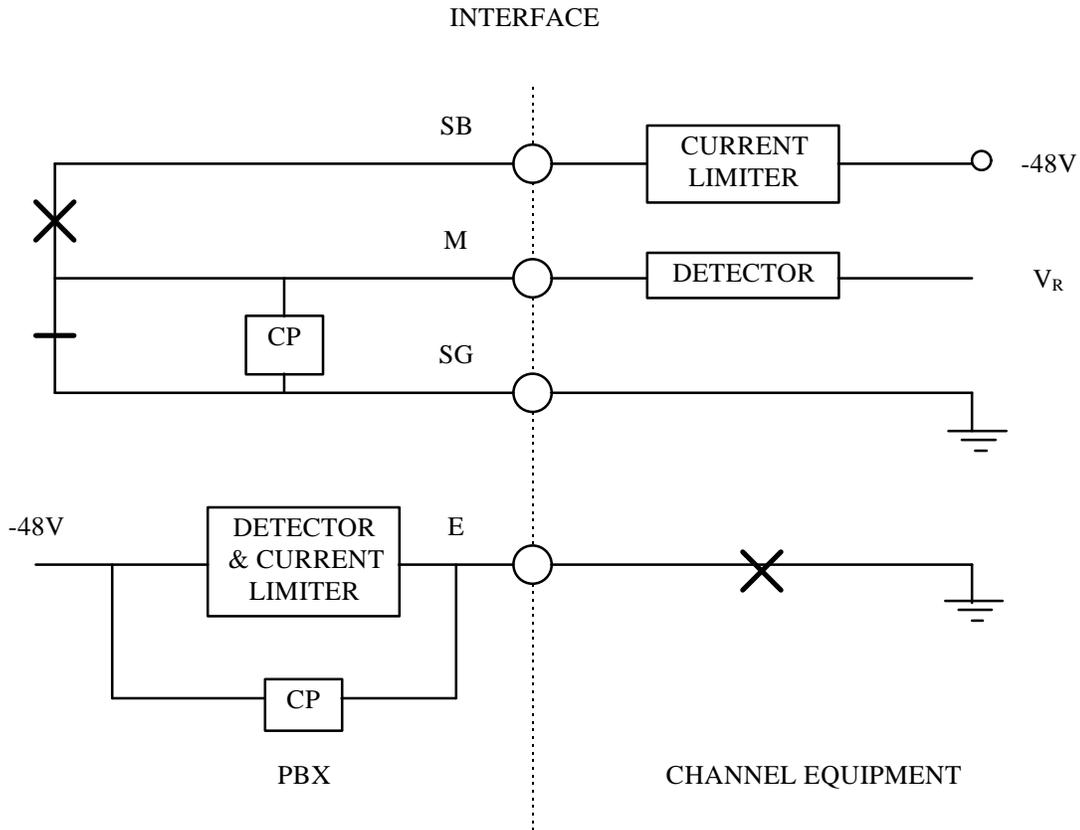
- a) No DC signals should be applied to the T, R, T or R₁ conductors.
- b) Electrical requirements for E&M Type III E-lead circuits are identical to those for Type I lead circuits, as specified in EIA/TIA-464-B.
- c) If the M lead switching component is a relay, then the following recommendations apply:
 - M lead transfer contact protection should be provided by limiting voltage transients during contact transfer time (conventionally this has been done for mercury relay and wire-spring contacts by insertion of a 1 K ohm, 5 W resistor from the outgoing side of the M lead to SG. A Zener diode of breakdown voltage of 65 V ± 10%, in series with a 1 K ohm, 1/2 W resistor, is now recommended as a general replacement for the 1 K ohm, 5 W resistor to reduce power consumption. The diode alone will suffice if the transfer time of the M lead pulsing contacts is no more than 1 ms., as happens with most wire-spring relays and mercury contact relays).
 - Break-before-make M lead transfer contacts should be used, with the break interval limited to a maximum of 1 or 2 ms.
- d) The M lead switching component should operate properly when connected to a signaling facility applying -42.5 to -52.5 volts on the SB lead.

2.2.4.2 On-Hook

- a) As an on-hook indication, the CPE should apply closure across the M and SG leads.
- b) When -50 V is applied to the M lead externally through a series resistance of 1000 ohms, the potential drop between the M and SG leads should not exceed 1 V in the on-hook state.

2.2.4.3 Off-Hook

- a) The CPE should apply closure across the M and SB to show the off-hook state.
- b) In the off-hook state, the potential drop from the M lead to SB lead should not exceed 2V with 50 mA in the M lead.
- c) The current in the SB lead should be equal to the M lead current, ± 10 percent. Any difference between the two lead currents implies incomplete separation of signaling and trunk circuit power systems, a condition that is contrary to the intent of the Type III interface arrangement.



X Contact closed for off-hook, open for on-hook

- Contact open for off-hook, closed for on-hook

CP Transfer contact protection

Figure III - 1. E&M Signaling Interface - Type III

2.3 T1.5 (DS-1)

Currently, all T1.5 (DS-1) special access facilities used for access to the services in this document will be designed for either Superframe (SF) or Extended SuperFrame (ESF) framing and channel format. Bipolar with 8-Zero Substitution (B8ZS) line coding is preferred, but may not be available on all access links. If B8ZS is not available, Alternate Mark Inversion (AMI) line coding is used instead. For voice and voiceband data services, mu-255 pulse code modulation (PCM) is used.

The reader should refer to AT&T Technical Reference TR 62411 for T1.5 interface specifications relating to:

- testing and maintenance.
- special network codes.
- alarm and status conditions.
- excessive bipolar violations.
- physical interface.
- electrical characteristics.
- jitter.
- keep-alive signals.
- loop-back.
- longitudinal balance.
- synchronization and timing.

If CPE is not loop timing off the incoming T1.5 service, or is not using a valid PRC derived source (Startum 1 traceable), timing slips on the T1.5 will occur. If the ACP receives excessive slips, the trunks will be maintenance disabled by the network. If an available source of network timing is present, it should be used. See AT&T Technical References TR 62411 and TR 60110 for digital network synchronization.

2.3.1 Information Encoding

This section contains specifications for encoding end-to-end user information (i.e., PCM voice or digital data). Supervisory signaling is covered in Section 2.3.2.

2.3.1.1 Pulse Code Modulation

AT&T Technical Reference TR 43801 provides a complete description of the SF framing, channel format and PCM recommendations. Also see ANSI standard EIA/TIA-464B for encoding loss and decoding loss recommendations.

2.3.1.2 56 Kbps Digital Data

In addition to voice calls, 56 Kbps, synchronous, circuit switched data calls are possible when the DS-1 interface is used. This capability, which is used by the SDN and Switched Digital 56 Services, is full duplex unless the 56Kbps digital broadcast capability is used (the 56 Kbps digital broadcast capability is described in Part I).

Each word (i.e., "octet") of an SF time slot consists of eight bits, numbered 1 through 8. Bit 8 is periodically robbed for signaling use. User data may occupy bits 1 through 7. To assure network compatibility and end-to-end compatibility (which may involve dissimilar CPE), the following recommendations should be met:

- a) Binary "1"s are represented by pulses and binary "0"s are represented by the absence of pulses .
- b) Once a data call is set up, the CPE should transmit a binary "1" in the eighth bit of all words of the relevant time slot (including words appearing in both signaling and nonsignaling frames).
- c) The CPE should always assure that no "all-zeros" word is sent. This constraint, known as the "per-channel pulse density constraint," is met while the call is in progress by observing requirement (b) of this section. During call set-up, or while the trunk is idle, any method of per-channel pulse density enforcement is acceptable, provided it does not interfere with normal framing, formatting, and supervisory signaling.
- d) By convention, bits of information appearing in a time slot word are ordered chronologically, with bit 1 of the word representing the earliest bit of user data, and bit 7 of the word representing the most recent bit of user data.

2.3.2 Supervisory Signaling

Two robbed bit channels (A and B) exist within SF framing and formatting, and four (A, B, C, and D) exist within ESF. The ACP uses the A/B-bits (SF) or A/B/C/D bits (ESF) in a fashion that is consistent with E&M channel units. Thus, only two supervisory states are used: "on-hook" and "off-hook." These are summarized below.

2.3.2.1 General

The CPE should support robbed bit signaling channels A and B, in both transmit and receive directions, in accordance with the SF framing and formatting standard, as specified in AT&T TR 43801. All new CPE designed for use on DS-1 special access facilities should also support the ESF robbed bit signaling channels (A, B, C, and D). AT&T recommends that CPE treat all signaling bits as described here, to assure signaling uniformity among equipment of differing types, and to avert possible future compatibility problems.

2.3.2.2 On-Hook

- a) The CPE must transmit signaling bits A=B=0 (SF) or A=B=C=D=0 (ESF) as an on-hook indication.
- b) The CPE must interpret a "0" in the received signaling bits as an on-hook indication from the ACP.

2.3.2.3 Off-Hook

- a) The CPE must transmit signaling bits A=B=1 (SF) or A=B=C=D=1 (ESF) as an off-hook indication.
- b) The CPE must interpret a "1" in the received signaling bits as an off-hook indication from the ACP.

2.4 56 Kbps Single Channel

The 56 Kbps single channel special access is currently used with both the SDN and Switched Digital 56 Services (see Part I for brief descriptions of these services).

The 56 Kbps single channel special access arrangement employs 56 Kbps DATAPHONE Digital Service (DDS) facilities from the AT&T serving office to the customer premises. The signaling conversion function in the AT&T serving office maps the robbed bit signaling present on the DS-1 facility into the Data and Control modes supported by DDS. The DDS facilities obtained from the LEC are otherwise a conventional, unmodified type, without secondary channel. AT&T Technical Reference TR 62310 describes the DDS interface, including the Data and Control modes used to convey the supervisory signaling.

Customer premises equipment shown in Figure II-1 includes a separate TIU, which adapts the DDS line coding to a standard DCE/DTE interface (such as V.35 or RS-449/422) and which performs the call control procedures described in Section 3 (the TIU may alternatively be integrated within the terminal equipment. In either case, all CPE is provided by the customer). The CPE is expected to loop-time off the received signal.

2.4.1 Information Encoding

User data is encoded onto the DDS loop as described in AT&T TR 62310. That document describes how data is encoded in a bipolar, return-to-zero format, with a positive or negative pulse representing a binary "1" and the absence of a pulse representing a binary "0." The CPE should send a Zero-code Suppression (ZS) bipolar violation sequence (see Figure II-2b) to show a string of seven consecutive binary "0"s in the user's data. Consequently, the CPE should never transmit more than six consecutive unsuppressed "0"s. Zero-code Suppression sequences are also transmitted by the network toward the customer. It is, however, possible that the network may transmit as many as twelve consecutive unsuppressed "0"s toward the CPE. Except for ZS sequences, no bipolar sequences are allowed once a call is established. Control Mode Idle (CMI) is not supported as an end-to-end signaling mechanism, since CMI is used for supervisory signaling. Only the 56 Kbps data rate is supported.

2.4.2 Supervisory Signaling

2.4.2.1 On-Hook

- a) As an on-hook indication, the CPE should transmit Control Mode Idle (CMI) bipolar violation sequences toward the network. Figure II-2a illustrates a typical CMI sequence.
- b) The CPE should interpret CMI bipolar violation sequences as an on-hook indication from the ACP.

2.4.2.2 Off-Hook

- a) As an off-hook indication, the CPE should transmit a Data Mode signal toward the network. A data mode signal is characterized by one of the following two attributes:
 - i. Absence from bipolar violations.
 - ii. Zero-code Suppression (ZS) sequences, as in Figure- II-2b.
- b) The CPE should interpret a data mode signal as an off-hook indication from the ACP.

2.4.3 Example Implementation

The following example is intended solely to illustrate use of the single channel access signaling. AT&T makes no claim that this implementation is or is not suitable for CPE, and does not discourage alternative implementations which may be more feature-rich and/or more cost effective.

A standard 56 Kbps CSU/DSU, intended for DDS use, may be used as a building block for a TIU.

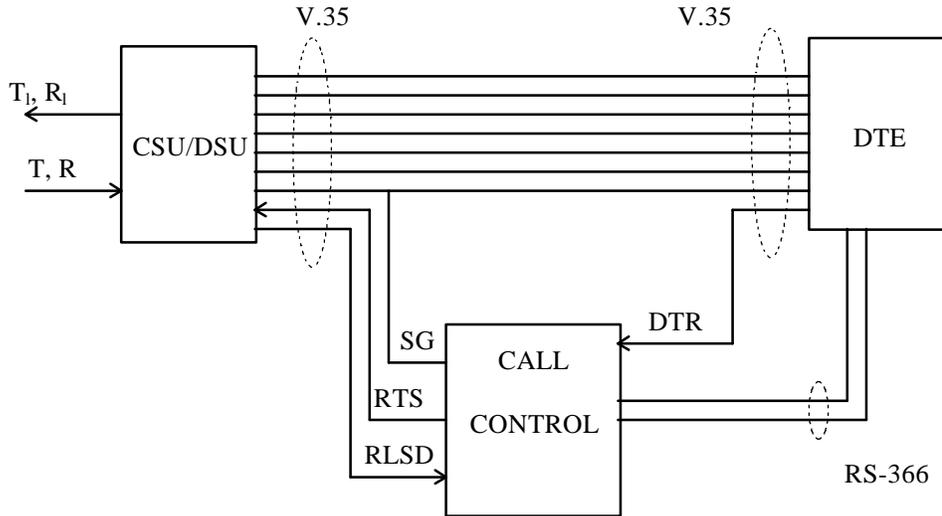


Figure III -2. Example TIU implementation

Figure III - 2 shows such an implementation. Normally, a DSU will transmit in Data mode whenever the Request To Send (RTS) lead is held true, and will transmit CMI whenever RTS is held false. Furthermore, a received Data mode signal is reflected on the Received Line Signal Detect (RLSD) lead as a true condition and received CMIs result in RLSD being false. Thus the outgoing supervisory state is controlled by RTS and the incoming state is evident by examining RLSD. Both of these signals are typically available on the DCE/DTE interface (which is usually a V.35 connector). The call controller, which communicates with the DSU over the RTS and RLSD leads, can be controlled remotely through a command port as illustrated in Figure III - 2 or can be controlled by a key pad. The Data Terminal Ready (DTR) lead may be used to inhibit the call controller from answering or originating calls when the DTE is off-line. Procedures executed by the call controller are described in Section

The DSU is strapped to derive transmit clock from the received line signal, and reports both received clock (RC) and transmit clock (TC) to the DTE. Data is sent and received over the transmit data (TD) and received data (RD) leads, respectively, as in a DDS installation.

3. CALL CONTROL PROCEDURES

This section includes call set-up and tear-down scenarios relevant to special access. It differs from the treatment of this subject presented in Part II in that this section makes recommendations which are applicable at the customer premises interface rather than at the AT&T point of termination. Moreover, the timing recommendations given in this section provide appropriate margins to allow for the distortions impressed on the signal by typical access facilities¹.

1. As stated previously, the information in Part III is based on certain assumptions about the access facilities. If these assumptions are not valid, then the recommended margins may not be appropriate.

It is possible to mix signaling types on a given trunk (e.g., DDS in with wink start outward), but some signaling combinations are not allowed. Table II-1 is a comprehensive list of signaling type combinations allowed on E&M and DS-1 trunks. Combinations allowed on 56 Kbps single channel special access trunks are listed in Table II-2.

Glare is the condition that exists when two offices (the ACP and the CPE) each seize the same two-way trunk at almost the same time. Glare detection and resolution are specific to each call control procedure, and are described in the relevant subsection. If a two-way trunk is shared by MEGACOM[®] and/or Toll Free MEGACOM[®], the ACP will resolve the glare. Otherwise, it is possible for a customer to choose whether the glare will be resolved by the CPE or by the ACP. If the CPE is able to detect and resolve glare, it is usually advantageous that it do so on trunks not shared by MEGACOM[®] and/or Toll Free MEGACOM[®]. Methods for detecting and resolving glare are given for wink start call control in Section 3.1.3 and for delay-dial / start-dial call control in Section 3.2.3.

For various causes, known or unknown, there may be short changes (called hits) in the received supervisory state. These may be either on-hook hits or off-hook hits. It is necessary that the CPE be reasonably immune to hits. Hits are typically of relatively short duration (compared with normal call control signals), and the CPE should ignore them on the basis of their short duration. Appropriate hit timing varies with circumstance, and is specified where applicable.

All flash timing should be handled within the CPE. No flashes should be presented to the ACP.

In the outward (i.e., from CPE) direction, all trunks use Direct Outward Dialing (DOD). In the inward direction, trunks may use Direct Inward Dialing (DID) or may suppress outpulsing.

3.1 Wink Start

Wink start is the preferred method of call control because it inherently offers an integrity check on the trunk, and because it provides for rapid detection and resolution of glare on two-way trunks.

3.1.1 Idle State

When the trunk is idle, both the CPE and the ACP are on-hook. The idle condition exists until either the CPE or the ACP seizes the trunk.

3.1.2 Call Set-Up

3.1.2.1 Trunk Seized by Calling CPE&A

The trunk is initially idle. The CPE seizes the trunk by changing its supervisory state to off-hook. The CPE keeps the trunk off hook (except during address transmission, if dial pulsing is used) until:

- a) the call is disconnected (if the call completes normally).
- b) the call is abandoned before completion.
- c) the CPE resolves a glare condition (assuming the CPE is the glare-resolving party).
- d) the CPE declares a wink time-out.

The ACP, upon recognizing the seizure and preparing to receive address information, sends a wink (i.e., a momentary, timed off-hook signal). On an active, properly functioning trunk, the CPE should expect to receive a wink almost immediately. If the CPE does not detect a wink within 5 seconds after seizure, it should declare a wink time-out. The CPE should be capable of recognizing a wink as short as 100 ms., but should consider an off-

hook signal shorter than 70 ms. to be a hit. An off-hook signal lasting 70 to 100 ms. may be interpreted either as a hit or as a wink, at the CPE vendor's option. If the CPE receives an off-hook signal persisting longer than 350 ms. on a two-way trunk, and it is administered to resolve glare, then it should declare a glare condition and resolve it as described in Section 3.1.3.

The CPE should delay transmission of address information for at least 70 ms. after detecting the trailing edge of the wink. The CPE transmits address information in the form of dial pulses (Section 4.1.1), dual tone multifrequency (Section 4.2.1), or multifrequency (Section 4.3.1) on E&M or DS-1 trunks². On 56 Kbps single channel access trunks, only dial pulse is allowed.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel. On E&M and DS-1 trunks, the CPE should apply these tones to the calling party's terminal.

If answer supervision to the calling CPE is provided, the ACP will transmit an off-hook signal toward the CPE to indicate that the far-end CPE has answered the call. The answer signal will last for at least 2 seconds. An off-hook excursion lasting less than 0.6 seconds should be interpreted as a hit.

Notes:

1. Off-hook hits are common on 56 Kbps single channel access trunks during the time the CPE is awaiting an answer signal. Vendors are advised that the 0.6 second hit timing is extremely important in CPE intended for use on such trunks.
2. If a 56 Kbps special access station initiates a call to a station served by a LEC's public switched digital service (PSDS), it is possible that the answer signal will be returned to the special access station before the data connection is clear. CPE should be prepared to wait up to 3 seconds after detecting answer supervision for a usable channel.
3. When certain older switching offices are involved at the egress end of calls to non-special access destinations, a momentary off-hook signal may be returned as the call is being established. This off-hook signal will reach the calling CPE after the address digits have been sent, while the CPE is awaiting answer. If the CPE responds too quickly, this signal may appear as an answer signal followed by a disconnect. If such symptoms are experienced, the CPE should be adjusted so an answer signal is not considered valid until it persists for 4 to 5 seconds.

Answer delay varies with application. On a voice call, the answer delay depends on the human response time. In this case, the CPE should not execute any time-out procedures; the calling party will normally hang up after deciding the destination party is not available to answer the phone. On a 56 Kbps data call, the called CPE will often answer automatically, without need for human interaction. Vendors of CPE for the 56 Kbps digital data capability are encouraged to provide an answer-delay timer, the duration of which can be adjusted by the customer. For a 56 Kbps call to a domestic destination, an answer delay of about 30 seconds appears appropriate. International call set-up times are longer than domestic call set-up times and may be highly variable. CPE for use with the international 56 Kbps data capability should therefore be equipped with an answer-delay timer which is adjustable to at least 90 seconds.

If the calling party hangs up before the call completes, or if the answer delay time-out expires, the CPE should return the trunk to an idle state.

2. Choice of address signaling type is made at the time the trunk service order is placed.

3.1.2.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook. The CPE should regard a received off-hook signal lasting at least 90 ms. as a trunk seizure. A received off-hook excursion lasting less than 35 ms. should be regarded as a hit.

Once the called CPE has detected a valid seizure from the ACP, it should prepare to receive address digits (if the trunk uses Direct Inward Dialing) and, within 150 ms., send a wink to the ACP over the seized trunk. The wink is a momentary off-hook signal with a duration of 140 to 290 ms.

On Direct Inward Dialing (DID) trunks, the CPE should not register any address digits during the first 30 ms. after sending the trailing edge of the wink. The CPE should be prepared to register dial pulses within 70 ms. after sending the trailing edge of the wink. At least 70 ms. after it receives the wink, the ACP will begin to transmit address digits. Address digits may be in the form of dial pulses (Section 4.1.2), dual tone multifrequency (Section 4.2.2), or multifrequency (Section 4.3.2)³. After receiving all the digits, the CPE connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.) and alerts the selected destination party.

On trunks with suppressed outpulsing, the CPE alerts the appropriate station (e.g., a PBX attendant position) and connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.).

When and if the destination party answers, the CPE cuts the call through, removes the call progress tone generator, and returns answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers. On trunks with suppressed outpulsing, it should not be returned sooner than 70 ms. after the trailing edge of the wink. The CPE should maintain this off-hook signal until the call is disconnected.

3.1.3 Glare Resolution

If the ACP seizes the trunk at about the same time as does the CPE, a glare condition exists. As explained in Section 3.1.2.2, the CPE expects the ACP to respond to a seizure by sending a wink. In a glare situation, however, the CPE detects an off-hook signal which persists too long to be a wink.

If the CPE is the glare-resolving party, then it should declare a glare condition if the ACP's off-hook signal persists longer than 350 ms. The CPE should then abort its call attempt by returning the trunk to the on-hook state and should prepare to receive address digits from the ACP. The ACP will interpret the off-hook to on-hook transition as an indication that it may begin sending address digits.

If the ACP is the glare-resolving party, then the CPE should not time the wink. It should simply continue to assert the off-hook state. When the ACP finally resolves the glare, it will return the trunk to the on-hook state. The CPE may now begin sending address digits.

3.2 Delay-Dial / Start-Dial

Delay-dial / start-dial is sometimes referred to as delay-dial with integrity check. Delay-dial / start-dial call control procedures are similar to those for wink start, except that different timing recommendations apply. In particular, the CPE should observe different glare timing if it is the glare-resolving party on a two-way trunk. Moreover, the recommendations on the duration of the delay-dial signal emitted by a CPE are different than the recommendations on the duration of a wink.

3. Choice of address signaling type is made at the time the trunk service order is placed.

3.2.1 Idle State

When the trunk is idle, both the CPE and the ACP are on-hook. The idle condition exists until either the CPE or the ACP seizes the trunk.

3.2.2 Call Set-Up

3.2.2.1 Trunk Seized by Calling CPE&A

The trunk is initially idle. The CPE seizes the trunk by changing its supervisory state to off-hook. The CPE keeps the trunk off hook (except during address transmission, if dial pulsing is used) until:

- a) the call is disconnected (if the call completes normally).
- b) the call is abandoned before completion.
- c) the CPE resolves a glare condition (assuming the CPE is the glare-resolving party).
- d) the CPE declares a delay-dial signal time-out.

The ACP, upon recognizing the seizure sends a delay-dial signal (i.e., an off-hook signal), followed by a start-dial signal (i.e., a return to the on-hook state). On an active, properly functioning trunk, the CPE should expect to receive a delay-dial signal immediately. If the CPE does not detect a delay-dial signal within 750 ms. after seizure, it should declare a delay-dial signal time-out. The CPE should be capable of recognizing a delay-dial signal as short as 100 ms., but should consider off-hook signals shorter than 70 ms. to be hits. If the CPE receives an off-hook signal persisting longer than 4 seconds on a two-way trunk, and it is administered to resolve glare, then it should declare a glare condition and resolve it as described in Section 2.3.

The CPE should delay transmission of address information for at least 70 ms. after detecting the start-dial signal. The CPE transmits address information in the form of dial pulses (Section 4.1.1), dual tone multifrequency (Section 4.2.1), or multifrequency (Section 4.3.1) on E&M or DS-1 trunks⁴. On 56 Kbps single channel access trunks, only dial pulse is allowed.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel. On E&M and DS-1 trunks, the CPE should apply these tones to the calling party's instrument.

If answer supervision to the calling CPE is provided, the ACP will transmit an off-hook signal toward the CPE to indicate that the far-end CPE has answered the call. The answer signal will last for at least 2 seconds. An off-hook excursion lasting less than 0.6 seconds should be interpreted as a hit.

Note:

1. When certain older switching offices are involved at the egress end of calls to non-special access destinations, a momentary off-hook signal may be returned as the call is being established. This off-hook signal will reach the calling CPE after the address digits have been sent, while the CPE is awaiting answer. If the CPE responds too quickly, this signal may appear as an answer signal followed by a disconnect. If such symptoms are experienced, the CPE should be adjusted so an answer signal is not considered valid until it persists for 4 to 5 seconds.

4. Choice of address signaling type is made at the time the trunk service order is placed.

Answer delay varies with application. On a voice call, the answer delay depends on the human response time. In this case, the CPE should not execute any time-out procedures; the calling party will normally hang up after deciding the destination party is not available to answer the phone. On a 56 Kbps data call, the called CPE will often answer automatically, without need for human interaction. Vendors of CPE for the 56 Kbps digital data capability are encouraged to provide an answer-delay timer, the duration of which can be adjusted by the customer. For a 56 Kbps call to a domestic destination, an answer delay of about 30 seconds appears appropriate. International call set-up times are longer than domestic call set-up times and may be highly variable. CPE for use with the international 56 Kbps data capability should therefore be equipped with an answer-delay timer which is adjustable to at least 90 seconds.

If the calling party hangs up before the call completes, or if the answer delay time-out expires, the CPE should return the trunk to an idle state.

3.2.2.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook. The CPE should regard a received off-hook signal lasting at least 90 ms. as a trunk seizure. A received off-hook excursion lasting less than 35 ms. should be regarded as a hit.

Within 150 ms. after the called CPE has detected a valid seizure from the ACP, it should send a delay-dial signal (i.e., it should go off hook). If Direct Inward Dialing (DID) is used, the CPE should also prepare to receive address digits.

On E&M or DS-1 trunks, the delay-dial signal should be at least 140 ms. in duration, and the start-dial signal should not occur earlier than 200 ms. from reception of the seizure signal. On 56 Kbps single channel access trunks, the delay-dial signal should be at least 900 ms. in duration. In any case, the delay-dial signal should never be longer than 4 seconds.

If DID operation is used, the CPE should send a start-dial signal (i.e., it should go back on hook) when it is ready to receive address digits, subject to the constraints on the delay-dial signal duration stated above. If outpulsing is suppressed at the ACP, the CPE should send the start-dial signal at least 200 ms. after detection of the seizure.

On DID trunks, the CPE should not register any address digits during the first 30 ms. after sending the start-dial signal, but should be prepared to register dial pulses within 70 ms. after it sends the start-dial signal. At least 70 ms. after receiving the start-dial signal, the ACP will begin to transmit address digits. Address digits may be in the form of dial pulses (Section 4.1.2), dual tone multifrequency (Section 4.2.2), or multifrequency (Section 4.3.2) on E&M or DS-1 trunks⁵. Address digits on 56 Kbps single channel access trunks will be in dial pulse form. After receiving all the digits, the CPE connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.)⁶ and alerts the selected destination party.

On trunks with suppressed outpulsing, the CPE alerts the appropriate station (e.g., a PBX attendant position) and connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.)⁷.

When and if the destination party answers, the CPE cuts the call through, removes the call progress tone generator, and returns answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers. On trunks with suppressed outpulsing, it should not be returned sooner than 70 ms. after the start dial signal. The CPE should maintain this off-hook signal until the call is disconnected.

5. Choice of address signaling type is made at the time the trunk service order is placed.

6. CPE for use on 56 Kbps single channel access trunks do not send call progress tones.

7. Ibid.

3.2.3 Glare Resolution

If the ACP seizes the trunk at about the same time as does the CPE, a glare condition exists. As explained in Section 3.2.2.1, the CPE expects the ACP to respond to a seizure by sending a delay-dial signal, followed by a start-dial signal. In a glare situation, the CPE will detect an off-hook signal which persists too long to be a delay-dial signal. In other words, the start-dial signal does not arrive in time.

If the CPE is the glare-resolving party, then it should declare a glare condition if the ACP's off-hook signal persists longer than 4 seconds. The CPE should then abort its call attempt by returning the trunk to the on-hook state and preparing to receive address digits from the ACP. The ACP will interpret the off-hook to on-hook transition as a start-dial signal and begin sending address digits.

If the ACP is the glare-resolving party, then the CPE should not time the delay-dial signal. It should simply continue to assert the off-hook state. When the ACP finally resolves the glare, it will return the trunk to the on-hook state. The CPE may now begin sending address digits.

3.3 Immediate Start

Because immediate start operation is not amenable to glare detection, it is never recommended that two-way trunks be provisioned for immediate start in both the inward and outward directions. Immediate start can be used in the inward direction on a two-way trunk with inward dialing suppressed, but the CPE should be the glare-resolving party⁸. Immediate start operation provides no trunk integrity checking during senderized call set-up, and therefore its use on senderized one-way trunks is recommended only if the CPE cannot accommodate wink start or delay-dial / start-dial operation. In the inward direction, stop-go operation is permissible with dial pulse signaling.

3.3.1 Idle State

When the trunk is idle, both the CPE and the ACP are on hook. The idle condition exists until either the CPE or the ACP seizes the trunk.

3.3.2 Call Set- Up

3.3.2.1 Trunk Seized by Calling CPE&A

The trunk is initially idle. The CPE seizes the trunk by changing its supervisory state to off-hook. The CPE keeps the trunk off hook (except during address transmission, if dial pulsing is used) until:

- a) the call is disconnected (if the call completes normally).
- b) the call is abandoned before completion.

The ACP maintains the on-hook supervisory state during this stage of the call set-up.

The CPE should delay transmission of address information for at least 70 ms. after initially seizing the trunk. The CPE sends address digits to the ACP in the form of dial pulses (Section 4.1.1), dual tone multifrequency (Section 4.2.1), or multifrequency (Section 4.3.1).⁹

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel. The CPE should apply these tones to the calling party's instrument.

8. This type of operation is common on 56 Kbps trunks; it is unusual on voiceband and voiceband data trunks.

9. Choice of address signaling type is made at the time the trunk service order is placed.

If answer supervision to the calling CPE is provided, the ACP will transmit an off-hook signal toward the CPE to indicate that the far-end CPE has answered the call. The answer signal will last for at least 2 seconds. An off-hook excursion lasting less than 0.6 seconds should be interpreted as a hit.

Notes:

1. When certain older switching offices are involved at the egress end of calls to non-Special access destinations, a momentary off-hook signal may be returned as the call is being established. This off-hook signal will reach the calling CPE after the address digits have been sent, while the CPE is awaiting answer. If the CPE responds too quickly, this signal may appear as an answer signal followed by a disconnect. If such symptoms are experienced, the CPE should be adjusted so an answer signal is not considered valid until it persists for 4 to 5 seconds.
2. If a 56 Kbps special access station initiates a call to a station served by a LEC's public switched digital service (PSDS), it is possible that the answer signal will be returned to the special access station before the data connection is clear. CPE should be prepared to wait up to 3 seconds after detecting answer supervision for a usable channel.

In general, the answer delay will depend on the human response time. If the calling party hangs up before the call completes, the CPE should return the trunk to an idle state.

3.3.2.2 Trunk Seized by ACP

The trunk is initially idle. The ACP seizes the trunk by changing its supervisory state to off-hook. The CPE should regard a received off-hook signal lasting at least 90 ms. as a trunk seizure. A received off-hook excursion lasting less than 35 ms. should be regarded as a hit.

On DID trunks, the CPE should be ready to receive address digits within 65 ms. after seizure of the trunk by the ACP. Unless stop-go operation is used, the CPE maintains a steady on-hook signal toward the ACP until the call is answered.

If DID stop-go operation is used, the CPE may interrupt dial pulsing from the ACP by sending a "stop" signal (i.e., by going off hook). The stop signal should be sent in the interdigital time, after receipt of the first, second, or third digit, and at least 65 ms. before the end of the interdigital time. If the stop signal occurs too late, the call attempt will be aborted at the ACP and appropriate messages will be sent to the CPE. When the CPE is ready to accept more digits, it sends a "go" signal by going on hook. The stop signal may not exceed 4 seconds, and only one stop-go is allowed per call.

After receiving all the digits, the CPE connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.) and alerts the selected destination party.

If outpulsing is suppressed, the CPE immediately alerts the assigned station (e.g., PBX attendant position) and connects the trunk to the appropriate call progress tone generator (e.g., busy tone, audible ring tone, etc.¹⁰).

When and if the destination party answers, the CPE cuts the call through, removes the call progress tone generator, and returns answer supervision to the ACP by going off hook. The answer signal should be returned within 500 ms. after the destination party answers, but not sooner than 200 ms. after the initial seizure from the ACP. The CPE should maintain this off-hook signal until the call is disconnected.

10. CPE for use on 56 Kbps single channel access trunks do not send call progress tones.

3.4 Dial Tone Start

Dial tone can be provided by the ACP to support non-senderized (cut-through) operation in the outward (from CPE) direction. On dial tone trunks, the ACP will accept either dial pulse or Dual Tone Multifrequency address signals, on a per-call basis. The ACP does not support dial tone start call control in the inward (to CPE) direction.

3.4.1 Idle State

When the trunk is idle, both the CPE and the ACP are on hook. The idle condition exists until either the CPE or the ACP seizes the trunk.

3.4.2 Call Set-Up

The trunk is initially idle. The CPE seizes the trunk by changing its supervisory state to off-hook. The CPE keeps the trunk off hook (except during address transmission, if dial pulsing is used) until:

- a) the call is disconnected (if the call completes normally).
- b) the call is abandoned before completion.

The ACP will transmit dial tone to the CPE upon recognition of the seizure.

The CPE should delay transmission of address information for at least 70 ms. after detecting the dial tone. Alternatively, the CPE may delay for at least 220 ms. after initially seizing the trunk. The CPE may then send address digits to the ACP in the form of dial pulses (Section 4.1.1) or Dual Tone Multifrequency (Section 4.2.1)¹¹.

While the call is being set up, standard in-band call progress information (e.g., audible ring tone, busy, reorder, etc.) may be present on the transmission channel. The CPE should apply these tones to the calling party's instrument.

If answer supervision to the calling CPE is provided, the ACP will transmit an off-hook signal toward the CPE to indicate that the far-end CPE has answered the call. The answer signal will last for at least 2 seconds. An off-hook excursion lasting less than 0.6 seconds should be interpreted as a hit.

Note:

1. When certain older switching offices are involved at the egress end of calls to non-special access destinations, a momentary off-hook signal may be returned as the call is being established. This off-hook signal will reach the calling CPE after the address digits have been sent, while the CPE is awaiting answer. If the CPE responds too quickly, this signal may appear as an answer signal followed by a disconnect. If such symptoms are experienced, the CPE should be adjusted so an answer signal is not considered valid until it persists for 4 to 5 seconds.

In general, the answer delay will depend on the human response time. If the calling party hangs up before the call completes, the CPE should return the trunk to an idle state.

11. The CPE may outpulse in either format, on a per-call basis; the ACP assigns the appropriate receiver (DP or DTMF) upon detection of the first digit received from the CPE.

3.5 Disconnection

Disconnect procedures for calling CPE differ slightly from those for called CPE. This section lists the procedures for each, with these procedures further categorized according to which CPE has initiated the disconnection.

3.5.1 Calling CPE Procedures

3.5.1.1 Calling CPE Hangs Up First

The calling CPE may disconnect the call by changing the trunk's supervisory state to on-hook. The calling CPE's on-hook signal (called a disconnect signal) should persist for at least 250 ms. Within 1050 ms. after the ACP recognizes the disconnect signal, it will go on hook toward the calling CPE. The ACP's on-hook signal is an indication that the call has been disconnected and the trunk is idle. The calling CPE should not attempt to originate another call on this trunk until the received on-hook signal has persisted for a guard time of at least 700 ms. Within 750 ms. (Note: the network could seize the trunk immediately after an on hook condition. This should be no problem unless CPE design does not tolerate incoming seizures during guard interval) after initiation of the ACP's on-hook signal, the calling CPE should be prepared to properly process a new incoming seizure.

3.5.1.2 Called CPE Hangs Up First

If the called CPE hangs up first, the call will not be disconnected (and the billing clock will not be stopped) until:

- a) the calling CPE disconnects as described above.
- b) a time-out period of up to 20 seconds elapses.

If the called CPE returns the trunk to an off-hook state before either of these events occurs, the call will not be disconnected. If answer supervision is provided to the calling CPE, the ACP will change its state to on-hook toward the calling CPE to indicate that the called CPE has hung up. Upon detection of such an indication, the calling CPE may opt to cut short the 20-second delayed disconnect by simply setting the trunk on hook for at least 250 ms.

3.5.2 Called CPE Procedures

3.5.2.1 Calling CPE Hangs Up First

If the calling CPE disconnects the call (as described in Section 3.5.1.1), the ACP will change the supervisory state of the egress trunk to on-hook. When it receives an on-hook signal of at least 300 ms. duration, the called CPE should interpret a disconnect, and should return the trunk to the on-hook state within 250 ms. A slower response may be interpreted by the ACP as a new seizure.

3.5.2.2 Called CPE Hangs Up First

If the called CPE hangs up first, the call will not be disconnected until:

- a) the calling CPE disconnects as described in Section 3.5.1.1. or,
- b) a time-out period of up to 20 seconds elapses.

If the called CPE returns the trunk to an off-hook state before either of these events occurs, the call will not be disconnected. The called CPE should therefore be prepared to wait up to 20 seconds before receiving confirmation that the call has been disconnected. When the call is disconnected, the ACP will confirm this by changing the

supervisory state of the trunk to on-hook. The called CPE should observe an on-hook signal of at least 300 ms. duration before declaring the trunk idle. On-hook excursions of shorter duration are hits.

4. ADDRESS DIGIT TRANSMISSION

Special access CPE may use Dual Tone Multifrequency (DTMF), Multifrequency (MF) or Dial Pulse (DP) addressing. To some extent, the addressing types may be intermixed on two-way trunks, but not all combinations are allowed. Moreover, some restrictions apply based on facility type and call control procedure type. Vendors should refer to Tables II-1 and II-2 for a comprehensive list of allowed combinations.

4.1 Dial Pulse

Dial pulse address signaling consists of a sequence of momentary on-hook excursions (breaks), the number of such breaks corresponding to the value of a dialed digit (the exception is the digit "0," which is represented by a sequence of ten breaks). The dial pulse break interval is the time interval corresponding to the on-hook state of the trunk during dialing. The dial pulse make interval is the time interval corresponding to the off-hook state of the trunk during dialing. A break interval and an adjacent make interval constitute a dial pulse cycle. The percent break is the proportion of the dial pulse cycle occupied by the break interval.

Speed is expressed in pulses per second (pps), and it refers to the pulse repetition rate averaged over the transmission of a single digit, not including interdigital time. Equivalently, it is the reciprocal of the dial pulse cycle time.

Interdigital time is measured from the trailing edge (final break-to-make transition) of the last pulse of one digit to the leading edge (initial make-to-break transition) of the first pulse of the following digit. During the interdigital time, the trunk is off hook.

4.1.1 Dial Pulse Generation

- a) The CPE should generate dial pulses with a percent break in the range of 58 to 64 percent (the break interval generated by the CPE to represent the digit "1" should be between 53 and 80 ms.).
- b) In generating a dial pulse digit, the CPE should pulse at a speed between 8 pps and 11 pps.
- c) When senderized operation is used, the first digit should be outpulsed between 70 ms. and 14 seconds after:
 - i. the detection of the start dial signal (on delay-dial / start-dial trunks).
 - ii. the detection of the trailing edge of the wink (on wink start trunks).
 - iii. initial seizure of the trunk by the CPE (on immediate start trunks, with or without dial tone).

If the ACP does not receive the first digit in time and the trunk remains off hook, it will give the trunk permanent signal treatment.

- d) When senderized operation is used, the CPE should send dial pulses with an interdigital time between 600 ms. and 15 seconds. A maximum interdigital time of 1 second is desirable.
- e) The total duration of short makes and breaks at the initiation of a make interval should not exceed 3 ms.
- f) Any spurious break other than those in item e) above, during any off-hook interval while the CPE is in the address signaling state, should not exceed 1 ms. in duration (by definition, the address signaling state is

entered upon detection of a start-dial signal or a wink on delay-dial / start-dial trunks or wink start trunks, respectively. On Immediate start trunks, the address signaling state is entered upon seizure of the trunk. The address signaling state ends 600 ms. after the final digit is outpulsed.)

- g) Spurious makes in the break interval of a dial pulse should not occur.

4.1.2 Dial Pulse Reception

- a) The CPE should recognize dial pulses within the speed range of 7.5 pps to 12 pps.
- b) The CPE should properly interpret dial pulses having a percent break between 40 and 80 percent.
- c) The CPE should accept interdigital times as short as 300 ms.
- d) The CPE should regard short makes and breaks within 3 ms. after the initiation of a make interval as hits.
- e) During the address reception state, the CPE should regard spurious break intervals of less than 1 ms. as hits (by definition, the address reception state is entered upon dispatch of a start-dial signal or a wink on delay-dial / start-dial trunks or wink start trunks, respectively. On Immediate start trunks, the address reception state is entered upon detection of trunk seizure. The address reception state ends 600 ms. after the final digit is received.)
- f) During the address reception state, the CPE should regard spurious make intervals of less than 1 ms. as hits.

4.2 Dual Tone Multifrequency

Dual tone Multifrequency (DTMF) address signaling is a method of signaling using the voice transmission path. In general, this method may use up to 16 distinct signals. Each signal is synthesized by superimposing two sinusoidal tones, one each from two geometrically-spaced groups of frequencies, designated the "low group" and the "high group." The ACP allows the use of 12 of these 16 signals to represent the digits 0 through 9, plus the octothorp ("#") and asterisk ("*") characters. Use of the alpha characters (A-D) is not allowed. Table II-3 summarizes the frequency pairs used to represent the allowed DTMF digits.

4.2.1 DTMF Digit Generation

4.2.1.1 Four-Wire Analog Trunks

Specifications for DTMF digit generation as stated in EIA/TIA-464-B are applicable. Type II DTMF receivers are not recommended.

4.2.1.2 DS-1 Digital Trunks

Specifications in EIA/TIA-464B are applicable, insofar as the PCM content present on a DS-1 time slot represents the analog waveforms generated by a DTMF transmitter driving a 4-wire analog trunk. Part II relates PCM encode levels to analog transmission levels appropriate for DTMF transmitters.

4.2.2 DTMF Digit Registration

4.2.2.1 Four Wire Analog Trunks

Specifications on Type I DTMF receivers are provided in EIA/TIA-464B.

4.2.2.2 DS-1 Digital Trunks

Specifications on Type I DTMF transmitters, as stated in EIA/TIA-464B are applicable, insofar as the PCM content present on a DS-1 time slot represents the analog waveforms detected by a DTMF receiver attached to a 4-wire analog trunk. Part II relates PCM decode levels to analog transmission levels appropriate for DTMF receivers.

4.3 Multifrequency

The multifrequency (MF) signaling arrangements make use of pairs of frequencies out of a group of six frequencies. These frequencies are 700, 900, 1100, 1300, 1500, and 1700 Hz. On special access trunks, MF signals may be used for called number address signaling, and calling number identification. Only the numerical signals (0-9) and the control signals KP and ST may be used by customer premises equipment. The allowed MF codes are summarized in Table II-4.

Multifrequency signaling is used with wink start or delay-dial / start-dial call control. Two-way trunks may not mix MF signaling in one direction with DTMF signaling in the other (see Table II-1).

4.3.1 MF Digit Generation

4.3.1.1 Four-Wire Analog Trunks

The CPE, upon detecting the wink or start-dial signal, should transmit the required MF signals. The start of outpulsing should be delayed as little as possible, but it is desirable that the delay be not less than 50 ms. under normal conditions. When outpulsing on 2-way trunks that are not arranged to avoid glare, or to satisfactorily resolve glare, the start of outpulsing should be delayed as little as possible, but not less than 200 ms.

The transmitter and its signal should meet the following recommendations:

- a) The two frequencies of each code should start and end within 1 ms. of each other.
- b) The power output per frequency should be $-7 \text{ dBm} \pm 1 \text{ dB}$, at the 0-dB TLP (measured at the CPE transmitting equipment).
- c) There should not be over 1.0 dB difference between the powers of the two frequencies of any code.
- d) The total power of extraneous signal components should be at least 30 dB below the power level of either frequency of the code.
- e) The KP signal length should be 90 to 120 ms.
- f) The ST and digit signal lengths should each be 58 to 75 ms.
- g) The interval between signals should be 58 to 75 ms. The frequencies should be within 1.5 percent of stated nominal values.
- h) The transmitter should have the same nominal impedance as the interface on which it is used (600 ohm, nonreactive on 4-wire analog trunks).
- i) The transmitter should have a longitudinal balance to ground at least equal to those required for voice transmission.

- j) During tone-off periods, transmitted power at any MF frequency should not exceed -58 dBm₀ at the network interface.

4.3.1.2 DS-1 Digital Trunks

Specifications in Section 4.3.1.1 are applicable, insofar as the PCM content present on a DS-1 time slot represents the analog waveforms generated by an MF transmitter driving a 4-wire trunk. Part II relates PCM encode levels to analog transmission levels appropriate for MF transmitters.

4.3.2 MF Digit Registration

4.3.2.1 Four-Wire Analog Trunks

MF receivers should meet the following recommendations:

- a) The receiver impedance should match that of the interface on which it is used (600 ohm, nonreactive on 4-wire analog trunks). Its single frequency return loss should be at least equal to that required for voice transmission.
- b) The receiver should respond to signal levels between 0 and -25 dBm per frequency measured at the network interface. Existing receivers may have a sensitivity of only -22 dBm, but new circuits should meet the -25 dBm requirement. The receiver should not respond if the signal level drops below -35 dBm per frequency.
- c) The receiver should not respond to address signals prior to being "unlocked" by receipt of a KP signal. Simulation of the KP signal by speech or other signal or noise source should not cause more than one lost call per 2500 operator-originated calls. If two or more consecutive KP signals are received, all those after the first one should be ignored.
- d) Once unlocked, the receiver should remain unlocked until it receives the ST signal.
- e) The receiver should respond to signals in which each frequency component duration is at least 30 ms. The receiver should respond to a KP signal that is at least 55 ms. long and may respond if the KP signal is from 30 to 54 ms. long. The two frequency components may be shifted in time relative to each other by as much as 4 ms.
- f) It is desirable that the receiver not respond to signals shorter than the recommendations in the preceding paragraph, and it is required that it not respond to signals in which the two components are not coincident for more than 10 ms. The receiver should recognize interpulse intervals as short as 25 ms. This interval is defined as the time during which no signal frequency component is above -35 dBm. It is desirable that the receiver bridge interruptions as long as possible, consistent with meeting the interpulse requirement. It is required that it bridge interruptions up to 10 ms. long after the minimum length signal has been received.
- g) The receiver should accept up to 10 digits per second.
- h) The receiver should check for the presence of two, and only two, valid frequency components in each pulse. If a pulse fails to meet this requirement, the call should receive reorder treatment.
- i) The receiver should tolerate pulses in which there may be as much as 6 dB difference in power levels of the two frequency components. It is desirable that even greater level differences be tolerated.

- j) MF receivers should meet the same level of noise tolerance (message circuit noise and impulse noise) as do DTMF receivers. These recommendations, and the method for measuring CPE compliance with them, may be found in EIA/TIA-464B. In addition, MF receivers should tolerate the following types and levels of power line induction with an error rate of not more than one in 25,000 frequency-pair signals:

60 Hz	81 dBmC0
180 Hz	68 dBmC0

- k) The receiver should tolerate 2A-B and 2B-A modulation products caused by intermodulation when MF pulses are transmitted over standard carrier facilities. The power sum of these modulation products is expected to be at least 28 dB below each frequency component level of the signals.
- l) The longitudinal balance to ground should be equal to that required for voice transmission .
- m) The receiver should accept MF signals in which the component tones deviate in either direction by 5 Hz more than 1.5 percent of their nominal frequencies.

4.3.2.2 DS-1 Digital

Specifications in Section 4.3.2.1 are applicable, insofar as the PCM content present on a DS-1 time slot represents the analog waveforms detected by a DTMF receiver attached to a 4-wire trunk. Part II relates PCM decode levels to analog transmission levels appropriate for MF receivers.

5. CPE TRANSMISSION RECOMMENDATIONS

This section summarizes recommendations on CPE transmission characteristics relevant to special access connections. These recommendations are consistent with EIA/TIA-464B.

5.1 CPE Insertion Loss

Transmission losses and levels should be tightly controlled to assure high-quality, uniform voice telephone service. Typically, some of the end-to-end loss is budgeted to the CPE. EIA/TIA-464B provides appropriate CPE insertion losses for most applications.

Part II of this document defines the Digital Reference Signal (DRS) to be any sequence of PCM words which would be decoded by a zero level mu-255 decoder into a 0 dBm sinusoid with a nominal frequency of 1004 Hz. (the zero level mu-255 decoder is defined according to a deterministic set of PCM words, the "Digital Milliwatt.").

Decoder loss (DL) is defined as existing between a specified point on a digital portion (i.e., before PCM decoding) and a specified point on an analog portion of a telephone connection. The digital portion may, for example, be the receive pair of the DS-1 special access trunk interface (D/TO), and the analog portion might be an on-premises station line interface (ONS), close to the switch. If DRS is used as a test signal applied at the digital portion, then the signal observed at the analog portion will be a (nominal) 1004 Hz sinusoid of power level P_{ana} . Under these conditions, the decoder loss is defined to be

$$DL = -P_{ana}$$

where P_{ana} is expressed in dBm.

Similarly, encoder loss (EL) is defined as existing between a specified point on an analog portion and a specified point on a digital portion (i.e., after PCM encoding) of a given telephone connection .

To measure encoder loss, a zero level decoder is placed at the digital portion (e.g., D/TO). If the analog output power of the zero level decoder is P_{dig} and the signal power measured at the analog portion (e.g., ONS) is P_{ana} , then

$$EL = P_{\text{ana}} - P_{\text{dig}}$$

where P_{ana} and P_{dig} are both expressed in dBm.

With these definitions in mind, EIA/TIA-464B provides ELs and DLs which should be inserted by the CPE between various trunk and line interfaces. Encode loss (EL) for DTMF, MF, and call progress signal tone generators should be 0. Decode loss (DL) for DTMF and MF tone receivers should be 0.

5.2 Other CPE Transmission Requirements

EIA/TIA-464-B provides transmission recommendations which are appropriate for 2-wire and 4-wire special access CPE. Such recommendations apply to the following CPE characteristics:

- Frequency response.
- Dynamic range.
- Intermodulation distortion.
- Envelope delay difference.
- Return loss.
- Background noise.
- Impulse noise.
- Balance.
- Crosstalk.
- In-band signal power.
- Out-of-band signal voltage.
- Impedance.

Also, see AT&T TR 50150⁽¹⁾ for issues concerning the tandeming of calls and conference bridges.

6. CIRCUIT TESTING AND MAINTENANCE

Troubles should be reported by the customer to the appropriate service maintenance center, as determined by the application service. This center will then be in charge of testing and repairing the circuit as necessary. The customer is responsible for detecting failures of one-way outgoing (from CPE) trunks.

6.1 Preservice Testing of Circuits Provisioned on Voice-Grade Access Service

During the preservice testing, the actual circuit parameters are measured and recorded. Preservice testing is divided into two categories: transmission and operational.

6.1.1 Transmission Testing

Minimal preservice transmission testing verifies voiceband circuit performance on the basis of the following attributes:

- Continuity
- Signaling
- 1004 Hz loss
- Idle channel noise
- 404 Hz loss
- 2804 Hz loss

6.1.2 Operational Testing

After completing the preservice transmission tests, the associated group of trunks is placed in service at the ACP and the CPE is brought on line. The port-to-port losses associated with off-premises station (OPS) ports are specified in EIA/TIA-464-B. The operational testing is performed by exercising the CPE, ACP, and special access facilities as a total system, in a live simulation of service usage. The operational testing requires craft people at the customer premises and at the AT&T serving office.

6.1.3 Test Vehicle

The customer may assign a line appearance within the CPE to a 56A-like responder or other equipment. The recording of the line appearance will permit the AT&T technician to test the special access circuits through the Network Interface and into the CPE. This permits trunk testing without need for customer interaction. Alternative arrangements (which are less desirable) may require customer interaction or an on-site maintenance call.

6.2 Fault Location on DS-1 Access Lines

A telephone number for the Central Test Center will be provided by AT&T at the time of service installation. Testing and repair by AT&T can most efficiently be performed when a complete and accurate description of the difficulty is provided by the customer. Information such as the trouble, time of day, duration, etc. should be provided in reporting any trouble.

In addition, when problems are encountered by the customer, it is essential that the customer check the CPE for proper operation (while maintaining the keep-alive signals per AT&T TR 62411) prior to reporting the trouble. This action will eliminate unnecessary dispatches of repair personnel by both AT&T and its local access provider. For such dispatches, the customer will be required to pay a maintenance or service charge if the problem is isolated to the customer's equipment. Diagnostic capabilities which may be part of the customer's equipment can also be useful in trouble isolation. It is recommended that customer personnel be acquainted with the basic equipment layouts and visual indicators on the CPE, and be knowledgeable in fundamental maintenance techniques.

If the CPE is removed from service for this (or any other) reason, it is the customer's responsibility to provide a suitable keep-alive signal per AT&T TR 62411. The recommended keep-alive signal is an unframed pattern of all 1s or the AIS alarm, which the network recognizes as a "Red" alarm signal.

6.2.1 DS-1 Level Loop Backs

Loop-back (LB) provides the Network with the capability to perform single-ended fault location on the DS-1 facility. In-band loop-back is achieved through the application of a framed in-band control signal which indicates to the critical circuitry in the CPE that the receive signal should be looped back to the transmit pair. The specifications for this signal are as follows:

1. The CPE should enter the loop-back state and latch in that state upon the receipt of 5 seconds of a pulse sequence consisting of repetitions of a "1" (i.e., pulse) followed by four "0"s.
2. The CPE should remove the loop-back upon receipt of 5 seconds of a pulse sequence consisting of repetitions of a "1" followed by two "0"s.

While in a loop-back state, the CPE should not remove bipolar violations.

6.2.2 DS-0 Level Loop-Backs

Vendors are encouraged (though not required) to support a DS-0 level loop-back capability.

6.2.2.1 PCM-Bearing Trunks

On trunks used for voiceband applications, the customer may assign a trunk appearance to a 56A-like responder, as with the analog facilities.

6.2.2.2 Digital Data Bearing Trunks

CPE should also provide for loop-back on trunks used for 56 Kbps digital data. These loop-backs should respond to the maintenance codes discussed in this section, which originate at a test position within the Network (these maintenance codes are typical of those used for testing the DDS network). CPE should not attempt to send these codes to the distant CPE; such an attempt would disconnect the call.

Historically the 56 Kbps maintenance code set was defined based on the assumption that the DDS plant contained an Office Channel Unit (OCU), Channel Service Unit (CSU), and a Data Service Unit (DSU). The OCU mapped the maintenance codes present on the DS-1 portion of the circuit into a format appropriate for the 56 Kbps transmission facilities. The mapping of the maintenance codes which are relevant to the 56 Kbps single channel special access is summarized in Table III - 1.

Table III - 1. OCU (SA-DP) translation of Loop Back Codes

DS-0 LEVEL LOOP-BACK CODES		
DS-1 Mapping	56 Kbps Loop Mapping	Description
N0101100	Simplex current reversal	Channel loop-back
N0101010	Not applicable	OCU loop-back
N0101000	N0B0X0V	DSU loop-back

In the current regulatory environment, the assumptions about the DDS plant, as described above, are not valid in all cases. For example, the OCU function may be implemented in a Special Access Data Port (SA-DP) which is located on the customer's premises, and the CSU & DSU functions may be integrated within the TIU as illustrated in Figure 11-1. In light of this uncertainty, CPE vendors must use engineering judgment in deciding how these loop-back codes will be used most effectively in the environment for which their products are intended.

A DS-0 level loop-back should be operated upon receipt of a minimum of four (4) consecutive bytes of one of the following loop-back codes, remain operated as long as every other byte contains the loop-back code, and for a minimum of four (4) consecutive bytes after receipt of the last loop-back code:

N0101100 (DSU Loop-Back)

This code should actuate a loop-back at the DCE/DTE interface. It verifies proper operation of the digital communication equipment on customer premises. Such equipment may include, for example, a channel bank plus the data port and TIU.

N0101010 (OCU Loop-Back)

This code should actuate a loop-back at the point where the DS-1 signal is demultiplexed. This loop back requires proper operation of only a minimum amount of CPE circuitry. Typically, this code is implemented in a data port.

N0101000 (Channel Loop-Back)

If the CPE converts the DS-0 signal to a DDS format as an interim step before converting to a standard DCE/DTE format, the channel loop-back is appropriate. This loop-back tests the on-premises DDS-like line, but involves a minimum of circuitry at the station.

In the above loop-back codes, the "N" indicates that the state of this bit is uncertain and should be ignored. Figure III - 3 illustrates these loop-backs and the point at which they operate.

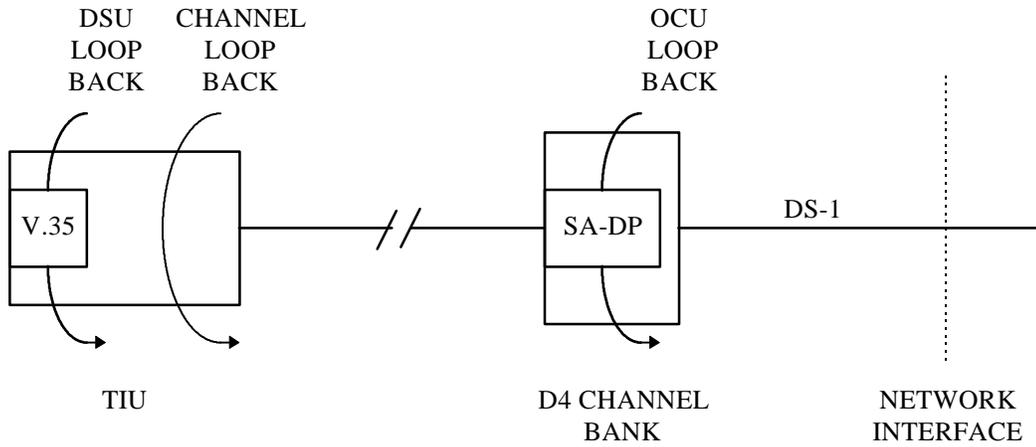


Figure III -3. Loop-back points in typical 56 Kbps special access CPE.

After loop-backs are established, all returned bytes (both data and refresh loop-back codes) should have the eighth bit set to 1. Accordingly, incoming loop-back codes should be mapped to produce a return byte as indicated in Table III - 2.

Table III -2. Alteration of echoed refresh loop-back codes

REFRESH LOOP-BACK CODES	
Loop-Back Code	Return Byte
N0101100	N0101101
N0101010	N0101011
N0101000	N0101001

6.3 Fault Location On 56 Kbps Digital Access Lines

Two signals intended to result in loop-back of the customer's received signal to the transmit side of the channel may be utilized. Implementation of loop-back circuitry in customer-provided terminal equipment allows remote testing of the DDS access circuit and to some extent isolation of trouble conditions between the Network and customer equipment. Response by the customer-provided terminal equipment to the first loop-back control signal described below is mandatory. Response to the optional loop-back control signal is recommended.

During call set-up, while the ACP continues to assert an on-hook indication toward the CPE, PCM-encoded call progress signals may be transmitted toward the CPE (these call progress signals are not intended for use by single-channel access CPE). Some of the resulting PCM words may be indistinguishable from DDS maintenance codes (including, but not limited to those described in this document), and in turn will be translated by the OCU (or SA-DP) and advanced toward the CPE. These "spurious" maintenance codes are generally isolated (i.e., do not repeat themselves in consecutive bytes). CPE should guard against them by requiring at least four consecutive bytes of the code before taking the indicated action.

6.3.1 Mandatory Channel Loop-Back

The mandatory loop-back test for single channel digital access uses reversal of the local cable simplex polarity to signal a loop-back test to the station apparatus. In the normal polarity, the transmit pair (T_1 , R_1) is kept positive with respect to the receive pair (T,R). Reversal of this polarity should be sensed in the simplex termination to control the channel loop-back. The sensing circuit should respond to the minimum 4 mA current described in TR 62310. To accomplish this mandatory loop-back, it is necessary for the station apparatus to perform full equalization and filtering of the received signal. The loop-back path should return this signal to the line driving circuitry of the transmitter for shaping in accordance with the recommendations of TR 62310. This loop-back should be designed to minimize penetration into the customer-provided terminal equipment.

6.3.2 Optional Channel Loop-Back (DSU Loop-Back)

The optional loop-back control signal is used to loop back the customer-provided terminal equipment at a point of logical separation between customer's data communications equipment and customer's terminal equipment. The optional loop-back state should be entered when 3 successive bipolar violation codes of the form N0B0X0V are received (N may be a zero or one). During testing, the signal will consist of alternating bytes of loop-back code and test bytes. The loop-back should be terminated upon receipt of 5 successive byte intervals without the loop-back indication. Implementation of this optional loop-back is highly recommended.

REFERENCES

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- 1 AT&T Technical Reference, TR 50150, AT&T Public Switched Telephone Network Changes to Transmission Characteristics, February 1996.