

# CONTENTS

	Page
PREFACE .....	viii
<b>PART I—TRUNKING</b>	
INTRODUCTION TO TRUNKING .....	I-ii
<b>CHAPTER 1: OVERVIEW OF TELEPHONE NETWORKS</b>	
Introduction .....	1-1
Circuits, Trunks, and Networks .....	1-1
Types of Trunks .....	1-7
CO Trunks .....	1-8
FX Trunks .....	1-8
WATS Trunks .....	1-8
Tie Trunks .....	1-8
DID Trunks .....	1-8
OPX Lines .....	1-9
Signaling .....	1-9
Supervisory States .....	1-9
Address Signaling .....	1-9
Information Signaling .....	1-12
Alerting Signals .....	1-13
Test Signals .....	1-14
Outpulsing Methods Used in Address Signaling .....	1-14
A Typical Telephone Call .....	1-14
CHAPTER 1 REVIEW QUESTIONS .....	1-17
<b>CHAPTER 2: OVERVIEW OF CIRCUIT OPERATION</b>	
Introduction .....	2-1
Introduction to Transmission Facilities .....	2-1
Metallic Transmission Facilities .....	2-1
Analog Carrier Transmission Facilities .....	2-3
Introduction to Signaling Systems and Formats .....	2-4
DC Signaling Systems .....	2-4
Loop Signaling .....	2-4
E&M Signaling .....	2-4
Duplex Signaling .....	2-4
Tone Signaling Systems .....	2-5
In-Band Tone Signaling .....	2-5
Out-of-Band Tone Signaling .....	2-5
Interfaces .....	2-5
Demarcation Point .....	2-5
4-Wire Termination Set .....	2-6
E&M Signaling Interface .....	2-6
End-to-end Change of State .....	2-9
Specific Signaling Formats .....	2-10
Loop Start .....	2-10
Ground Start .....	2-10
Delay Dialing .....	2-13
Wink Start .....	2-14
Direct Inward Dialing .....	2-14
CHAPTER 2 REVIEW QUESTIONS .....	2-15

**CONTENTS**  
(Continued)

**CHAPTER 3: LOOP START SIGNALING**

Introduction .....	3-1
Loop Start Overview .....	3-1
Idle Condition .....	3-2
Outgoing Call .....	3-2
Incoming Call .....	3-2
Call Disconnect .....	3-2
Detailed Operation of Loop Start Signaling .....	3-3
Idle Condition .....	3-3
Outgoing Call .....	3-4
Incoming Call .....	3-5
Loop Start to E&M Interface .....	3-6
Detailed Operation of Loop Start to E&M Interface .....	3-6
Outgoing Call (PBX Origination) .....	3-8
Incoming Call (CO Origination) .....	3-12
CHAPTER 3 REVIEW QUESTIONS .....	3-14

**CHAPTER 4: GROUND START SIGNALING**

Introduction .....	4-1
Ground Start Overview .....	4-1
Idle Condition .....	4-2
Outgoing Call .....	4-2
Incoming Call .....	4-2
Call Disconnect .....	4-3
Detailed Operation of Ground Start Signaling .....	4-3
Idle Condition .....	4-4
Outgoing Call, Closed End .....	4-4
Outgoing Call, Open End .....	4-6
Incoming Call, Open End .....	4-6
Call Disconnect .....	4-7
Ground Start to E&M Interface .....	4-8
Detailed Operation of Ground Start to E&M Interface .....	4-8
Outgoing Call (PBX Origination) .....	4-9
Incoming Call (CO Origination) .....	4-12
Call Disconnect .....	4-15
CHAPTER 4 REVIEW QUESTIONS .....	4-17

**CHAPTER 5: CONTROLLED OUTPUTSING—DELAY DIALING AND WINK START OPERATION**

Introduction .....	5-1
Delay Dialing .....	5-1
Wink Start .....	5-3
Glare .....	5-5
CHAPTER 5 REVIEW QUESTIONS .....	5-7

# CONTENTS

(Continued)

## CHAPTER 6: DIRECT INWARD DIALING

Introduction .....	6-1
DID Overview .....	6-1
Detailed Operation of DID Signaling .....	6-2
Idle Condition .....	6-2
Incoming Call, CO End .....	6-2
Incoming Call, PBX End .....	6-3
Call Disconnect .....	6-3
CHAPTER 6 REVIEW QUESTIONS .....	6-5

## APPENDIX A: OVERALL LOOP START OPERATION

Operation of Loop Start Signaling .....	A-1
Idle Condition .....	A-2
Outgoing Call (PBX Origination) .....	A-2
Incoming Call (CO Origination) .....	A-6
Call Disconnect .....	A-8
Operation of Loop Start to E&M Interface .....	A-10
Outgoing Call (PBX Origination) .....	A-10
Incoming Call (CO Origination) .....	A-17

## APPENDIX B: OVERALL GROUND START OPERATION

Operation of Ground Start Signaling .....	B-1
Idle Condition .....	B-2
Outgoing Call (PBX Origination) .....	B-3
Incoming Call (CO Origination) .....	B-8
Call Disconnect .....	B-12
Operation of Ground Start to E&M Interface .....	B-13
Outgoing Call (PBX Origination) .....	B-13
Incoming Call (CO Origination) .....	B-20

## APPENDIX C: OVERALL DIRECT INWARD DIALING OPERATION

Operation of Direct Inward Dialing (DID) Signaling .....	C-1
Idle Condition .....	C-2
CO Seizure .....	C-2
Start Dialing .....	C-2
Dial Pulsing .....	C-4
PBX Answer .....	C-4
Disconnect .....	C-5

## APPENDIX D: ANSWERS TO REVIEW QUESTIONS

Chapter 1 .....	D-1
Chapter 2 .....	D-1
Chapter 3 .....	D-2
Chapter 4 .....	D-2
Chapter 5 .....	D-2
Chapter 6 .....	D-2

**CONTENTS**  
(Continued)

**PART II—TRANSMISSION**

	Page
<b>INTRODUCTION TO TRANSMISSION</b> .....	ii-ii
 <b>CHAPTER 1: CHARACTERISTICS OF HUMAN SPEECH AND HEARING</b>	
<b>Introduction</b> .....	1-1
<b>Sound and Sound Waves</b> .....	1-1
<b>Pure and Complex Sounds</b> .....	1-3
<b>Speech</b> .....	1-4
<b>Hearing</b> .....	1-5
<b>Voice Frequency Band</b> .....	1-6
<b>CHAPTER 1 REVIEW QUESTIONS</b> .....	1-7
 <b>CHAPTER 2: UNITS OF MEASUREMENT IN TRANSMISSION</b>	
<b>Introduction</b> .....	2-1
<b>Milliwatt and Hertz</b> .....	2-1
<b>The Decibel</b> .....	2-3
<b>Decibel Measurement Relative to One Milliwatt</b> .....	2-4
<b>Transmission Level Point</b> .....	2-4
<b>Noise Measurement Units</b> .....	2-7
<b>dBrn</b> .....	2-7
<b>dBrnC</b> .....	2-7
<b>dBrnC0</b> .....	2-8
<b>CHAPTER 2 REVIEW QUESTIONS</b> .....	2-9
 <b>CHAPTER 3: TRANSMISSION PERFORMANCE PARAMETERS</b>	
<b>Introduction</b> .....	3-1
<b>Loss</b> .....	3-1
<b>Signal Distortion</b> .....	3-3
<b>Attenuation Distortion</b> .....	3-3
<b>Frequency Shift Distortion</b> .....	3-4
<b>Delay Distortion</b> .....	3-4
<b>Noise</b> .....	3-4
<b>Idle Channel Noise</b> .....	3-5
<b>Signal-Dependent Noise</b> .....	3-6
<b>Impulse Noise</b> .....	3-7
<b>Crosstalk</b> .....	3-7
<b>Echo</b> .....	3-7
<b>Balance Theory</b> .....	3-9
<b>Definition of Terms</b> .....	3-10
<b>Balance Testing</b> .....	3-10
<b>CHAPTER 3 REVIEW QUESTIONS</b> .....	3-12

**CONTENTS**  
(Continued)

**CHAPTER 4: TELCO INTERFACES**

Introduction .....	4-1
2-Wire Interface .....	4-1
2-Wire with E&M Interface .....	4-1
4-Wire with E&M Interface .....	4-2
CHAPTER 4 REVIEW QUESTIONS .....	4-3

**CHAPTER 5: TRANSMISSION TESTING**

Introduction .....	5-1
Actual Measured Loss .....	5-1
Test Configuration .....	5-1
Test Procedures .....	5-2
Calculating the Results .....	5-3
Slope .....	5-3
Test Procedures .....	5-3
Calculating the Results .....	5-3
Idle Channel Noise .....	5-4
C-Notched Noise .....	5-4
Test Procedures .....	5-4
Calculating the Results .....	5-4
Balance .....	5-4
Calculating the THL .....	5-5
Measuring the THL .....	5-5
Calculating the Results of the THL Test .....	5-7
Measuring the ERL .....	5-7
Calculating the Results of the ERL Test .....	5-7
Measuring the SRL .....	5-7
Calculating the Results of the SRL Test .....	5-8
Evaluating the Balance Test Results .....	5-8
Adjusting Balance .....	5-8
CHAPTER 5 REVIEW QUESTIONS .....	5-9

**APPENDIX A: ANSWERS TO REVIEW QUESTIONS**

Chapter 1 .....	A-1
Chapter 2 .....	A-1
Chapter 3 .....	A-1
Chapter 4 .....	A-2
Chapter 5 .....	A-3

INDEX .....	I-1
-------------	-----

**TRUNKING AND TRANSMISSION FINAL EXAM**

**ANSWER KEY TO TRUNKING AND TRANSMISSION FINAL EXAM**



## **PREFACE**

Since the first telephone conversation between Alexander Graham Bell and Dr. Watson, telephonic communication has expanded into a massive system that serves most of the world.

In the United States, history of the development of telephony was largely the history of the Bell System until the late 1960s. Since then, changes in the regulatory environment and rapid technological development have led to a diversification of the resources available for satisfying growing corporate telecommunications needs. It has become economically advantageous for many corporations (as well as operating companies in the "public" network) to take advantage of newly available transmission facilities, switches and associated equipment.

With the growth of corporate (or "private") networks containing components from a multitude of vendors, responsibility for network installation and maintenance has shifted from the "telephone company" to the corporation itself.

Of the many disciplines involved in building and maintaining networks, trunking and the associated subjects of signaling and transmission are the primary concerns of this book. The intent is to provide entry level technicians with the theoretical background that is prerequisite to the understanding and application of the practical discipline of trunk testing and maintenance. The level of presentation assumes a knowledge of basic electricity.

The book is divided into two parts: Part I provides an overview of networks and trunks, and details several types of signaling operation; Part II is devoted to the transmission characteristics of trunks. Review questions designed to reinforce concepts covered in the text follow each chapter, and a final exam is provided at the end.

In addition to providing a self-taught course on trunking, this book is intended to serve as a reference during a technician's further training and professional career.





**PART I**  
**TRUNKING**



## **INTRODUCTION TO TRUNKING**

Part I of this book will provide the technician with an overview of trunking. The chapters are broken down as follows:

Chapter 1 presents an overview of telephone networks.

Chapter 2 discusses the types of transmission facilities used in telephone networks and the signaling systems used over those facilities.

Chapter 3 presents an overview of loop start operation as well as a detailed discussion of loop start signaling with reference to block diagrams.

Chapter 4 presents an overview of ground start operation as well as a detailed discussion of ground start signaling with reference to block diagrams.

Chapter 5 contains a detailed discussion of delay dialing and wink start operation.

Chapter 6 provides a functional overview of Direct Inward Dialing (DID) signaling, a detailed discussion of the DID states in a DID call, and a discussion of overall DID signaling operation with reference to simplified circuit schematics.



# **CHAPTER 1**

## **OVERVIEW OF TELEPHONE NETWORKS**

### **INTRODUCTION**

The purpose of this chapter is to present an overview of telephone networks. The reader should gain an understanding of the components of a network and how they are related to one another, as well as an understanding of the basic terminology used in telephony.

The most fundamental transmission path in a telephone network is a *circuit*. A *line circuit*, or loop, is a single circuit that is dedicated to a *station set*.

*Trunks* consist of one or more *trunk circuits* and serve to connect switching machines. An essential difference between a trunk and a line circuit is that a trunk is a common connection *shared* by many users.

There are many types of trunks, and they are usually defined with reference to the type of switching equipment they connect as well as the function they perform within the system.

The types of trunks that are described in this chapter are:

- Central Office (CO)
- Foreign Exchange (FX)
- Wide Area Telecommunications Service (WATS)
- Tie Line (TIE)
- Direct Inward Dialing (DID)

In addition to the above trunks, a commonly used line circuit, the Off-Premise Extension (OPX), is described. Many of the considerations that apply to trunks also apply to OPX lines.

Switching machines are connected by trunks to form a switched network that is accessed by individual station sets over individual *circuits* or *loops*.

The overall telephone system is composed of the public network and private networks. Switching in private networks is generally provided by *Private Branch Exchanges* (PBXs).

*Signaling* is the process of sending control information over the network to control the setup, holding, charging and releasing of connections or communication pathways through the network.

Signaling may be divided into five functional categories:

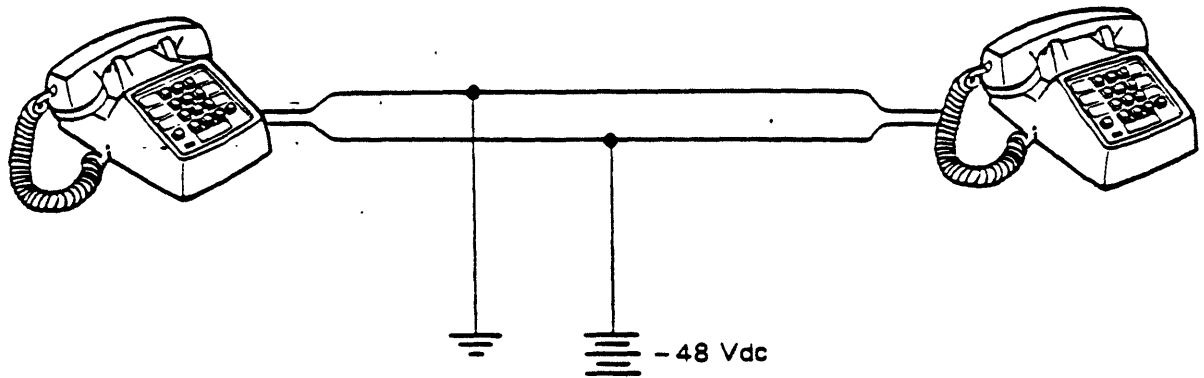
- Supervisory
- Address
- Information (Call Progress)
- Alerting
- Test

The essential functions of signaling are illustrated by following the progress of a typical call. Each functional category of signaling is then discussed in detail.

### **CIRCUITS, TRUNKS, AND NETWORKS**

A telephone network can be regarded as a systematic development of interconnecting transmission media arranged so that one telephone user can talk to any other user in that network. The most elementary telephone "network" consists of a single circuit connecting two telephones. This network, illustrated in Figure 1-1, requires two telephones, a circuit, a power source and a means of alerting each subscriber of a call from the other.

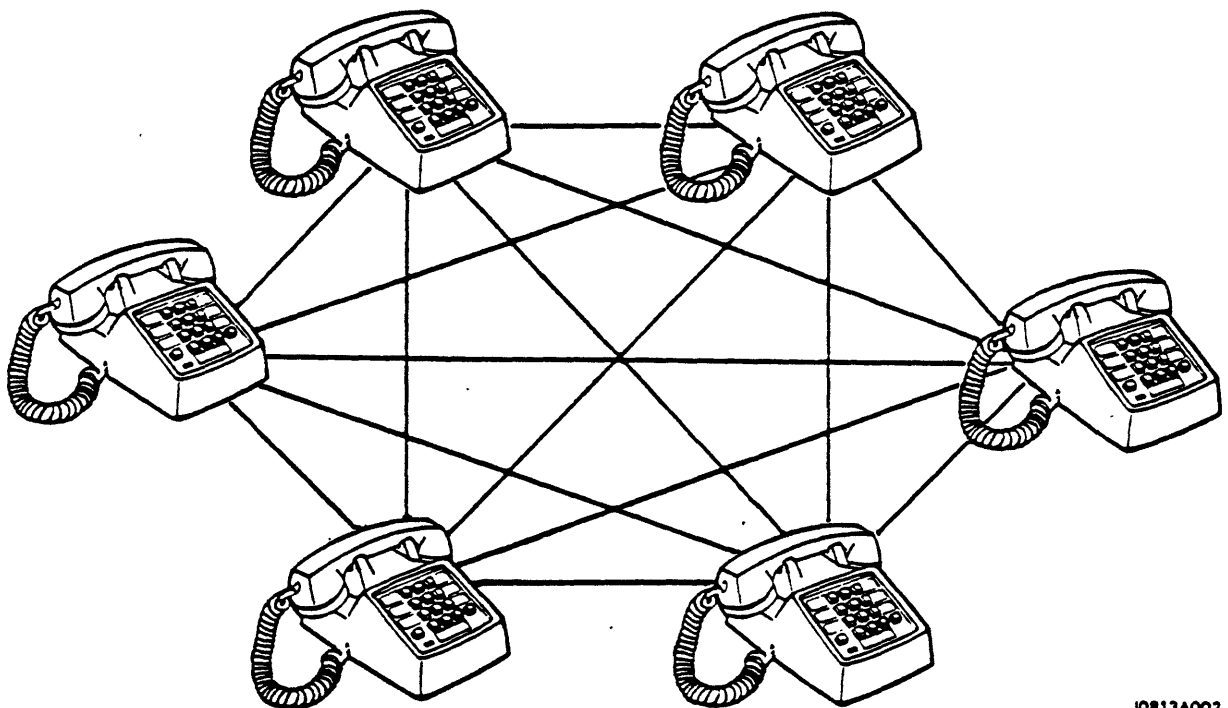
**Part I  
Trunking**



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**Figure 1-1. A Simple Telephone Connection**

A straightforward method of expanding this network in order to serve new subscribers is to provide a circuit between each pair of subscribers. A rudimentary network which connects six homes in this manner is illustrated in Figure 1-2.



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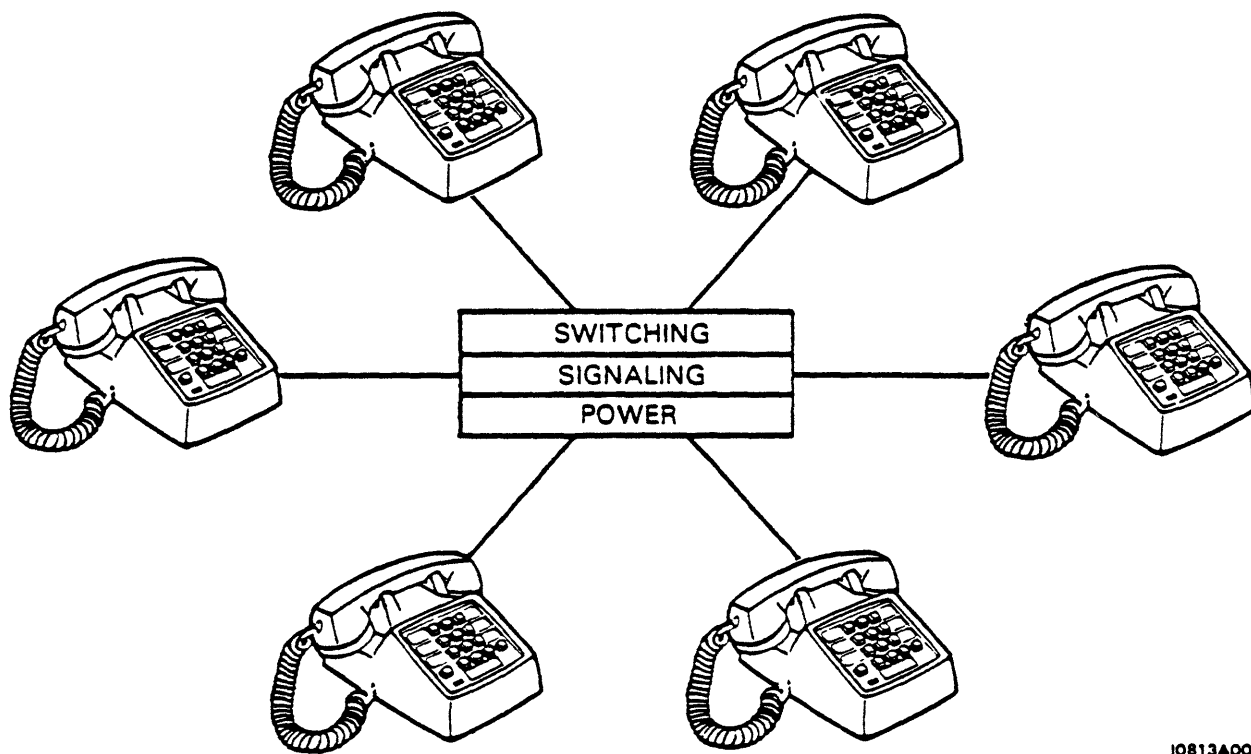
**Figure 1-2. A 6-Point Mesh Network**

To provide full access to each phone by every other phone, this network configuration requires 15 circuits to connect the homes. In addition to the telephone instrument, each home needs a power source, a switching device to direct a call to the desired destination, and some means of alerting the destination subscriber of an incoming call.

Clearly, this arrangement has several disadvantages. A large number of long wires is needed, and the number required increases dramatically as the network grows. To fully connect  $N$  homes in this fashion,  $N(N - 1) \div 2$  two-wire pairs would be required. To add a new phone to the network,  $N$  additional wire pairs would have to be added and connected to the  $N$  existing phones. In addition, the switching device in every existing phone would have to be modified to accommodate the new phone. The mesh network requires a large amount of equipment, much of which will seldom be used. A more economical arrangement, commonly known as a star network, is illustrated in Figure 1-3. Switching, signaling and power functions are now located centrally (Central Office, or CO).

Instead of connecting every phone to every other phone, only a single circuit is required to connect an individual phone to the Central Office. Where 15 long lines were required, we now need only 6 shorter lines. The 6 individual switching devices have been replaced by the centrally located switch. This adds some complexity to the use of the network since the user must be provided with a way to operate the switch remotely. Because the switching apparatus is centrally located, it may be combined and shared among the users, since all of the users will not need to use it at one time. In a similar manner, power is centrally provided and the means to *signal* subscribers is shared among users.

The switch illustrated in Figure 1-3 might be part of the public network located in a Central Office, or it might be part of a private or corporate network located on the customer's premises and called a *Private Branch Exchange (PBX)*.



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**Figure 1-3. A Star Network**

## Part I Trunking

The practice of sharing equipment among many users in a network is essential to providing economical telephone service. Sharing is possible because each user requires use of the network for only a small percentage of a typical day. Much of telephone network design is concerned with balancing the service objectives (grade of service) of individual users against the requirements economy

The earliest centrally located switching devices were manually operated switchboards controlled by operators who received verbal instructions from the customer (see Figure 1-4). The switching function is provided by the operator. Signaling is accomplished through the use of lamps indicating the status of each subscriber's line and the verbal dialogue between the subscriber and operator. The operator alerts called subscribers by applying ringing voltage (provided by the central power supply) to the appropriate line.

Connection between calling and called subscriber is by means of cords. Any of the cords may be used by different subscribers at different times. The cords, then, are *shared circuits*.

To improve the economy and speed of telephone networks, manual switchboards have, for the most part, been replaced by automatically controlled switches.

Now suppose that there are two groups of users, A and B, as illustrated in Figure 1-5

Most of the calls placed by users in group A are to other users in group A and the same is true for group B, however, subscribers in A do have occasion to call subscribers in B, and vice-versa.

Service between users in group A and those in group B may be provided by linking Switches A and B by a group of circuits. In the illustration, there are two circuits between Switch A and Switch B. Each circuit is *shared* among all of the users in A and B and is called a *trunk circuit* or a *tie line*. A trunk circuit may be defined as "a shared circuit connecting two switching systems"

For reasons of economy, trunk circuits having a common function are grouped, forming *trunk groups*

Now let's expand our network by adding another group of subscribers and another switch, as in Figure 1-6.

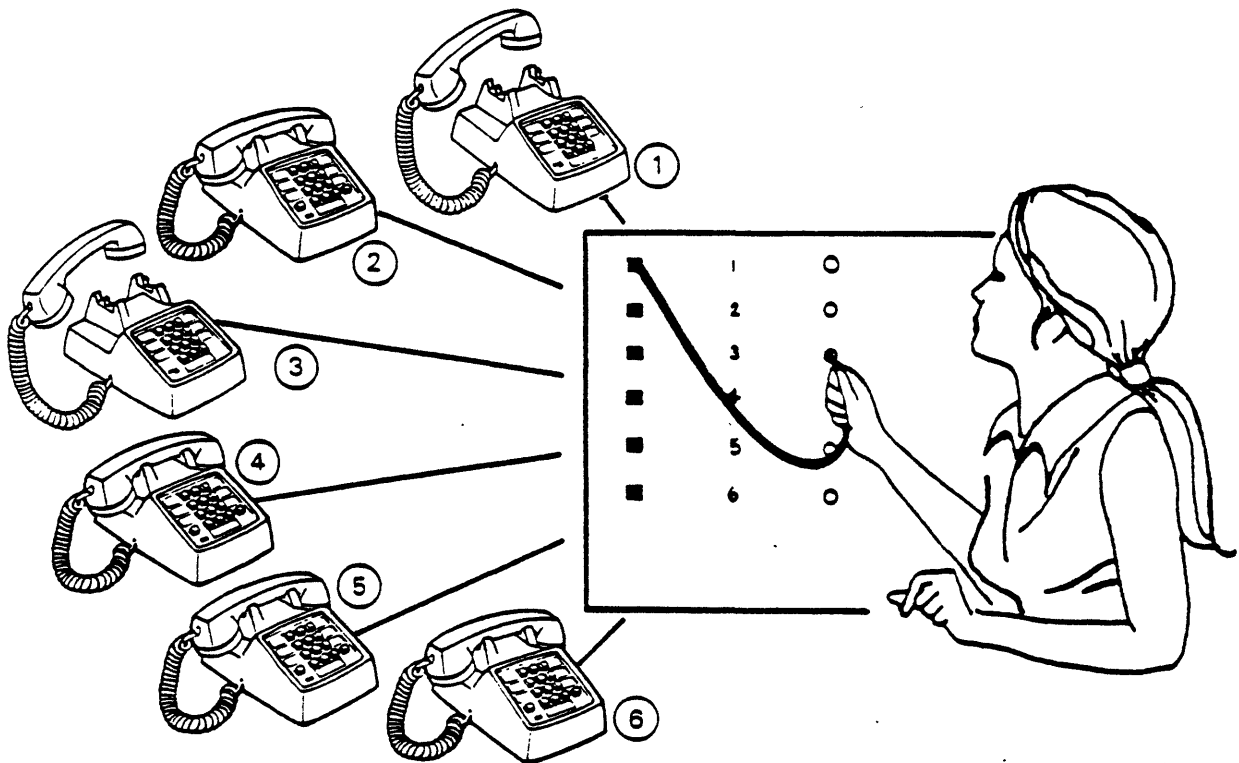
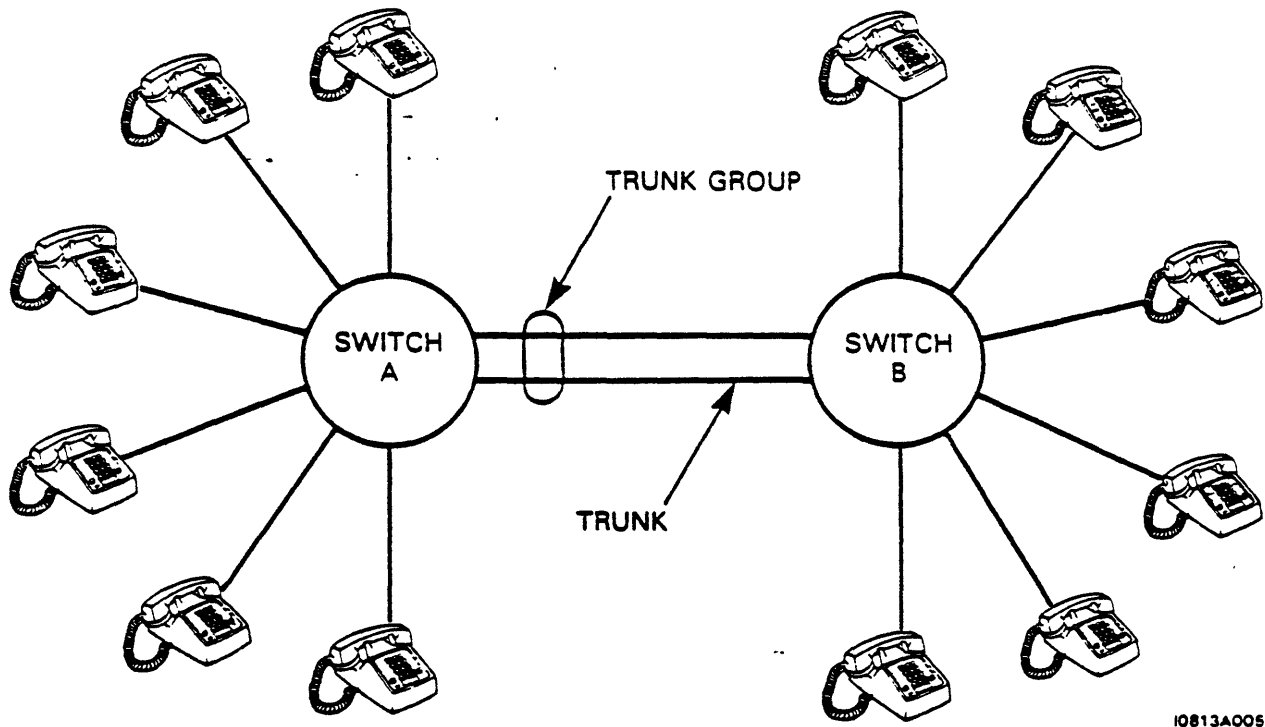


Figure 1-4. A Manually Operated Switchboard

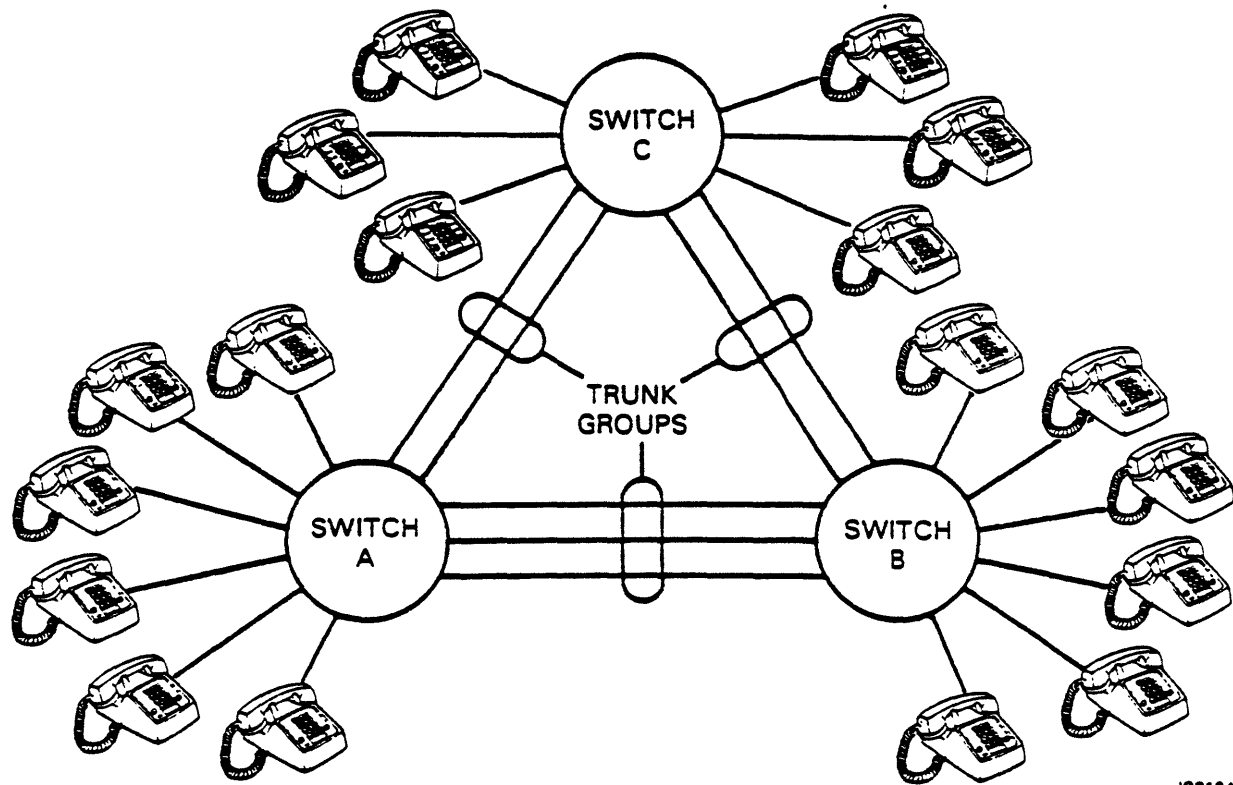
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Figure 1-5. A 2-Switch Network



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Figure 1-6. A 3-Switch Network

## Part I Trunking

There are two connecting paths between Switch A and Switch B: the path formed by trunk group AB and the path formed by trunk groups AC and CB, connected at Switch C. At any particular time, there may be no available circuit in trunk group AB. There may, however, be circuits available in trunk groups AC and CB. Networks are usually designed to take advantage of the different paths available for completing a call. This is called *alternate routing*, and its use can reduce the total number of circuits required in the network.

A subscriber in group A calling a subscriber in group B provides Switch A with the *address* of the desired subscriber. Based upon the address, Switch A will attempt to *route* the call directly to group B over trunk group AB, which is called the *first route choice*.

If there are no available circuits in trunk group AB, Switch A will attempt to send the call over trunk group AC, which is called the *second route choice* or, in this case, the *final route choice* (If no circuits are available in the final route choice, the switch will return a busy signal to the calling subscriber.) Once the call reaches Switch B, it is connected to the called subscriber, based upon the address information that has been passed to it by the intermediate switch or switches.

Let's summarize what we have discussed so far. A typical telephone network is composed of the following:

- Station Equipment (Telephones)
- Line Circuits
- Switching Equipment
- Trunk Circuits

Station equipment (the telephone set) is connected to switches by single circuits that are called line circuits or loops. Switching systems concentrate traffic from lines onto trunks and distribute traffic from trunk groups to lines. In addition, switching systems may have the ability to alternately route traffic. Efficient utilization of trunk groups is based upon the proper application of the concentration, distribution and alternate routing capabilities of switching systems.

The above discussion applies to private or corporate networks as well as to the familiar public telephone network. The first private network probably consisted of two telephones connected by a single circuit or loop, and was probably not connected to the public network. The first switches in private networks were manually operated switchboards through which individuals in a company could be connected with one another without traversing the public network. In addition, users could direct the operator to connect them with the public network via trunks specifically provided for that purpose.

Private networks have seen an evolution that is similar to that of the public network, with the additional requirement that private networks must connect with the public network in order to provide complete service. Although the services provided by point-to-point private lines and private networks could be approximated by using services available on the public network, private installations offer a number of advantages to the user. Where traffic is heavy between two geographical areas, private lines may be the most economical alternative. Fixed charges (not usage-sensitive) associated with private lines may be attractive economically. Also, privately controlled switching systems, particularly modern *Computerized Branch Exchanges* (CBXs), offer numerous features and services that provide convenience and range of use that may not be available on the public network. Of primary importance is the control of rising telecommunications costs that is provided by private networks.

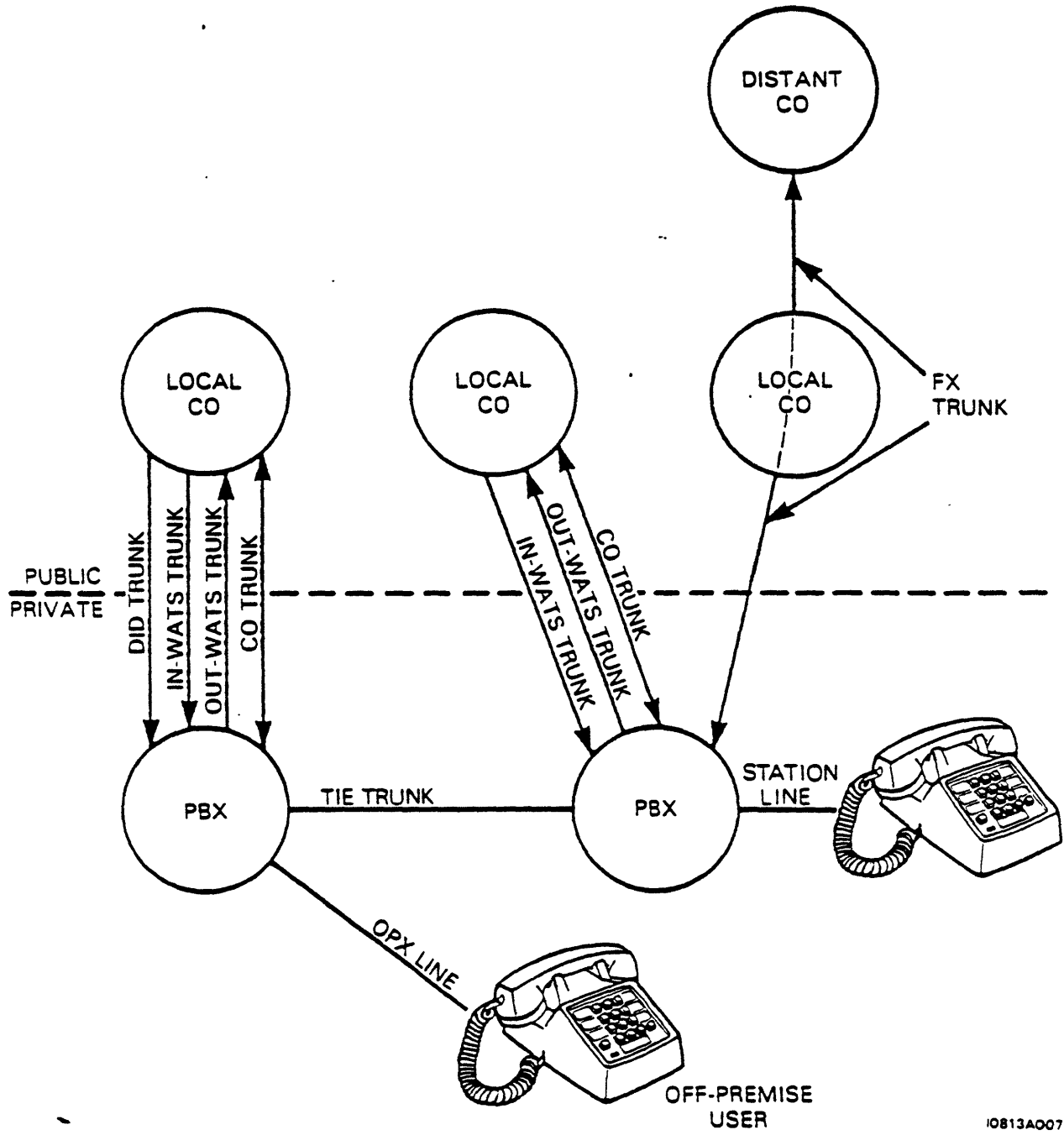
To summarize, private networks offer advantages to the corporation in the areas of specialized functions, convenience, economy and cost control. Private networks are usually constructed of components (station equipment, loops, trunks and switches) that are provided by, and perhaps owned by, a variety of different suppliers. With the increasing trend toward deregulation of the telecommunications industry, corporations are assuming greater control over their telecommunications systems. A primary purpose of this book is to provide the basic knowledge required to maintain circuits and trunks that make up private networks and their connection to the public network.

**TYPES OF TRUNKS**

Modern telephone networks are composed of a variety of different types of lines, trunks and switches. Figure 1-7 illustrates a small network segment consisting of both public and private sections.

The switching system illustrated in the private section is a *Private Branch Exchange (PBX)*. Only the lowest level of the public switched network, the Central Office (or CO), is illustrated here as it is sufficient to illustrate the types of trunks that will be discussed.

All trunks are capable of carrying information in two directions; however, some trunks are set up to allow call originations from only one end. These trunks are called one-way trunks. Figure 1-7 indicates direction by arrows.



**Figure 1-7. A Typical Network Segment**

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## Part I Trunking

**CO TRUNKS:** Central Office (CO) trunks are groups of circuits that connect a PBX to the Central Office that serves the local exchange. Their purpose is to connect the private network with the public switched network.

**FX TRUNKS:** Foreign Exchange (FX) trunks provide a direct connection between local customer equipment and a distant CO. FX trunks are generally used to provide service to a relatively compact community of interest that is at some distance from the local area. FX trunks provide the convenience of 7-digit dialing as well as service to a distant area at a reduced cost.

**WATS TRUNKS:** Wide Area Telecommunications Service (WATS), provided by the Bell system, allows customers to make (*Outward WATS*) or receive (*Inward WATS*) long distance calls and have them billed at a bulk rate on a total-time basis rather than on a per-call basis. Inward WATS calls are billed to the called number. For an interstate WATS customer, the Continental United States is divided into five service areas or concentric bands, extending outward from (but not including) the customer's home state, which is called Band 0. Hawaii and Alaska are served by Band 6. Intrastate WATS is also available in most states. A customer may subscribe to some or all of the bands on either an inward or outward basis. Purchase of service to a distant band includes all of the bands closer to the home state.

The WATS service areas or bands are illustrated in Figure 1-8.

WATS service can result in substantially reduced costs, particularly when a customer's calling patterns tend to be geographically spread out.

**TIE TRUNKS:** Tie trunks are circuits that connect two PBXs. Tie trunks are generally leased for a fixed monthly fee and can provide cost savings over other types of service when traffic between the areas served by the PBXs is heavy.

**DID TRUNKS:** Direct Inward Dialing (DID) trunks allow a caller to dial directly a station or group of stations served by a PBX, without the necessity of the call being served by an attendant. The extension digits of the desired station are dialed by the customer and are passed to the PBX by the connecting CO.

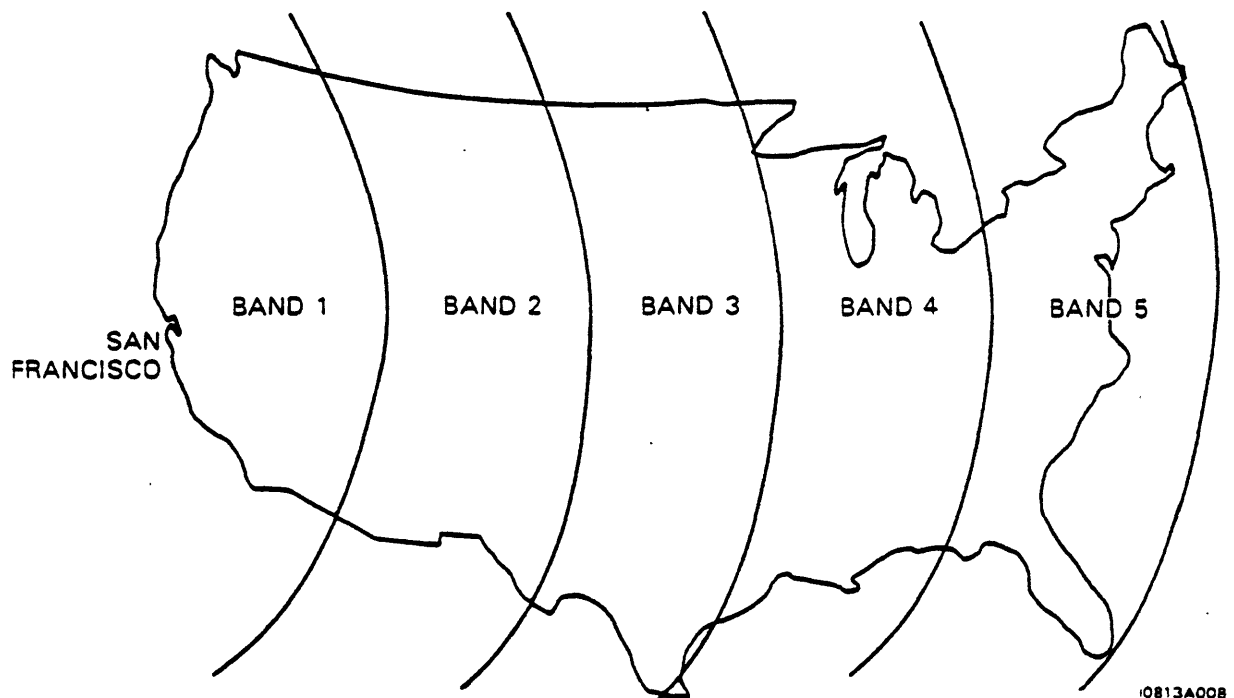


Figure 1-8. WATS Service Areas

**OPX LINES:** Off-Premise Extension (OPX) lines connect a distant (off-premise) extension to the PBX in such a way that the distant extension appears on site. Their purpose is to extend the convenience of extension dialing to telephones not located on the immediate premises.

## **SIGNALING**

Signaling in a telecommunications network is the exchange of control information between different functional parts of the system.

Signaling over circuits in a telephone network may be broken down into two areas:

- 1) The exchange of information between subscribers, both calling and called, and the switching machines; and
- 2) The exchange of information between switches in the switched network.

Subscriber line signaling has the primary function of initiating and terminating a call; it is essentially independent of the type of switching system and type of network.

Signaling between switches involves the transfer of information forward (from calling to called party) and backward (from called to calling party) over the network, with a primary purpose of setting up a voice path upon call initiation and clearing the path upon call termination.

Signaling over the switched network is usually more complex than subscriber line signaling because certain functions are required to ensure efficient operation of the network. Further complexity is introduced by the requirement that machines incorporating state-of-the-art technology must be able to "converse" with machines that have been in service for decades.

The various signaling functions that will be discussed in this book may be divided into five categories:

- Supervisory States
- Address Signaling
- Information Signaling
- Alerting Signals
- Test Signals

**SUPERVISORY STATES:** Supervision serves to detect or change the state or condition of a subscriber or network line.

For purposes of supervision, there are only two possible conditions: *on-hook* and *off-hook*. The on-hook condition applies when a telephone set is idle (the handset resting on the cradle, or an equivalent condition). Off-hook applies when the telephone set is in the active condition (the handset removed from the cradle, or an equivalent condition).

When a subscriber goes off-hook the subscriber line is said to be *seized* and changes the state or condition from *idle* to *active*. The change in condition from off-hook to idle at both ends of a line is termed a *disconnect* and changes the state of the line from active to idle.

**ADDRESS SIGNALING:** Address signaling is the means by which the telephone number of the called subscriber is supplied to the network by the calling subscriber and subsequently transmitted through the network. Four types of address signaling will be covered here:

- Dial Pulsing
- Dual Tone Multifrequency (DTMF) Signaling
- Multifrequency Pulsing
- Common Channel Interoffice Signaling

## Part I Trunking

*Dial pulsing* is a means of transmitting digital information from a subscriber's dial to the Central Office equipment. The numerical value of each digit in the dialed telephone number is determined by the number of on-hook intervals in a train of pulses. The on-hook intervals for each digit are separated by short off-hook intervals, and the digits themselves are separated by relatively long off-hook intervals. The on-hook signals do not interfere with the function of the supervisory disconnect signal since their duration is much shorter than the minimum on-hook interval that is recognized as a disconnect.

The momentary openings of the loop operate a relay at the Central Office. The digits received may directly operate switching equipment or may be relayed forward by an outgoing dial pulse repeater.

Percent break is the percentage of the total pulsing period that the signaling device (rotary dial or relay contact) is in the open or on-hook condition.

Modern customer dials are designed to a break rate of 58 to 64 percent and operate at a rate between 8 and 11 pulses per second. Figure 1-9 graphically illustrates dial pulsing.

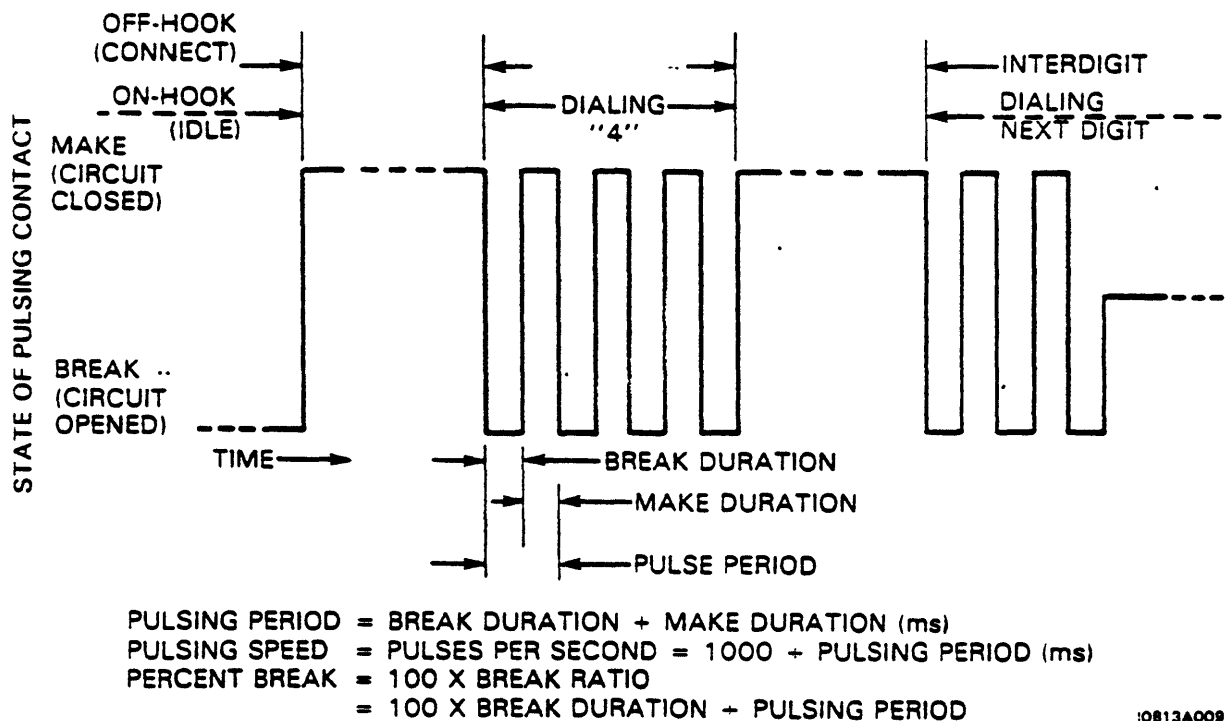


Figure 1-9. Dial Pulsing Definitions

*Dual Tone Multifrequency (DTMF) Signaling* provides a method for pushbutton signaling from customer stations over the voice transmission path. The DTMF code provides for 16 distinct signals. Each signal is composed of two voice band frequencies, one from each of two mutually exclusive frequency groups of four frequencies each.

Table 1-1 illustrates the frequency pairs that are used in DTMF signaling.

**Table 1-1. Frequency Pairs Used in DTMF Signaling**

LOW GROUP FREQUENCIES (Hz)	HIGH GROUP FREQUENCIES			
	1209	1336	1477	1633
697	1	2	2	Spare
770	4	5	6	Spare
852	7	8	9	Spare
941	*	0	#	Spare

Supervisory signaling is the same whether dial pulse or DTMF address signaling is used.

*Multifrequency (MF) pulsing* is a second type of signaling that uses frequency combinations rather than make-break pulse trains to transmit information. Unlike DTMF signaling, MF signaling is used between switches in the network rather than by the subscriber. The MF system is used instead of DTMF for interoffice signaling in order to avoid conflict between subscriber's address signaling and interoffice signaling. In addition, MF is less affected by long-haul transmission problems. The MF signaling system used principally in the United States and Canada is a two-out-of-five frequency pulse system. Additional signals for control functions are provided by combinations using a sixth frequency. Table 1-2 shows the digits or other applications and their corresponding frequency combinations as well as a brief explanation of "other applications."

**Table 1-2. Frequency Combinations**

DIGIT	FREQUENCY PAIR (Hz)
1	700 + 900
2	700 + 1100
3	900 + 1100
4	700 + 1300
5	900 + 1300
6	1100 + 1300
7	700 + 1500
8	900 + 1500
9	1100 + 1500
10 (0)	1300 + 1500

OTHER APPLICATIONS	FREQUENCY PAIR	EXPLANATION
KP	1100 + 1700	Preparatory for digits
ST	1500 + 1700	End of pulsing sequence
STP	900 + 1700	Used with TSPS (Traffic Service Position System)
ST2P	1300 + 1100	
ST3P	700 + 1700	
Coin collect	700 + 1100	Coin control
Coin return	1100 + 1700	Coin control
Ringback	700 + 1700	Coin control
Code 11	700 + 1700	Inward operator (CCITT No. 5)
Code 12	900 + 1700	Delay operator
KP1	1100 + 1700	Terminal call
KP2	1300 + 1700	Transit call

## Part I Trunking

*Common Channel Interoffice Signaling (CCIS)*, unlike previous address signaling methods, does not use the voice path to transmit signaling information. It is a system for exchanging information between processor-equipped switching systems over a network of separate signaling links. All signaling data, including the supervisory and address signals necessary to control call setup and takedown plus network management signals, will be exchanged by these systems over the signaling links instead of over the voice path, as is done using present signaling techniques. Figure 1-10 illustrates the difference between Common Channel Interoffice Signaling and more conventional signaling.

CCIS offers a number of important advantages over present inband signaling techniques. CCIS passes signals at higher speeds than conventional signaling systems; this allows calls to be set up and taken down more quickly. Network and trunk facilities may be used more efficiently because the holding time previously used for signaling is now available for carrying voice. CCIS signaling between processor-controlled switches is transmitted via high-speed digital bit streams, and a single signaling link may carry all of the signaling required for many transmission pathways. This sharing, as we have seen in other areas (switching and trunking), provides additional cost savings.

Call setup times now vary with the number of links in a connection. Due to the speed of CCIS, this variation will be reduced, resulting in more uniform call setup time regardless of the number of links involved.

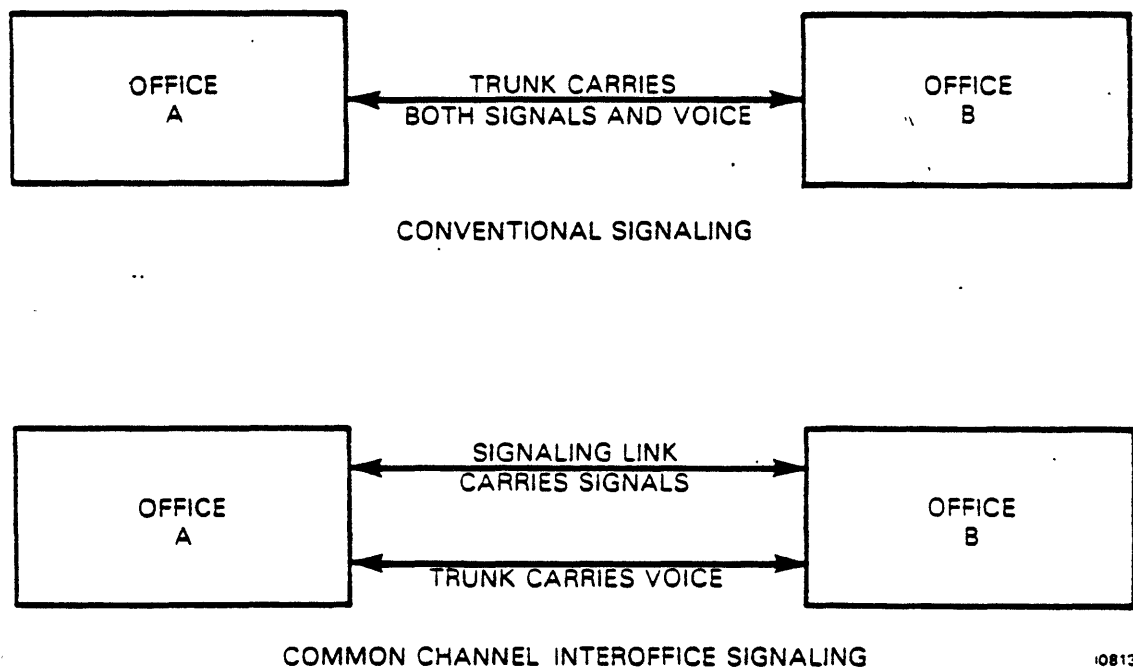


Figure 1-10. Common Channel Interoffice Signaling

**INFORMATION SIGNALING:** Signals in this category consist of audible tones that provide information regarding the progress or disposition of telephone calls to operators and subscribers.



As public and private networks evolved, several types of information tones and conventions applying to these tones were used. The advent of DTMF and MF address signaling created the possibility of conflict between tone-based address signaling and information signaling. To reduce the possibility of conflict, the Bell System adopted a *precise tone plan* which consists of four frequencies: 350, 440, 480 and 620 Hz. These tones are assigned singly or in pairs to represent standard audible tones described in the following paragraphs.

*Dial tone* consists of 350 Hz plus 440 Hz and is used to indicate that the exchange is ready to accept digits (either dial pulse or DTMF) from the subscriber. Once the first digit is dialed, dial tone is removed from the line.

*Audible ring* consists of 440 Hz plus 480 Hz and is returned to the calling subscriber to indicate that the called line has been reached and ringing has started.

*Line busy* is made up of 480 Hz plus 620 Hz, interrupted at 60 IPM, with approximately equal tone-on and tone-off times. Line busy indicates that the called line has been reached, but it is busy.

*Reorder* is alternately called paths busy, all trunks busy (ATB), or no circuit, and consists of 480 Hz plus 620 Hz, interrupted at 120 IPM. Reorder indicates that the local switching paths to the calling office or equipment serving the called customer are busy or that a toll circuit is not available.

The above audible-tone signals generally provide the subscriber with sufficient information regarding the progress or status of "routine" calls. *Recorded announcements* are used to provide additional information to notify the customer of an unexpected or abnormal condition that has prevented a call from completing.

Generally, the recorded announcements refer to conditions that exist either within the network or at the destination of the call. The announcements are worded so that the customer can understand the proper action to be taken. Under some conditions, the calling party will be answered by an operator after listening to the recording.

**ALERTING SIGNALS:** Alerting signals are used to notify a subscriber, operator or switching system of an existing condition or a request for action that requires immediate response.

The most familiar alerting signal is *ringing*, which is used to notify the called subscriber of an incoming call. Ringing is caused by the application of ringing voltage to the subscriber's line by the equipment co-located with the switching system (either a public network CO or a PBX). When audible ring is used, it is sent to the originating subscriber at the same time that ringing voltage is applied to the called subscriber's line.

*Flashing* refers to a momentary on-hook by a subscriber. The duration of the on-hook or "flash" is short and, therefore, is not mistaken for a disconnect. Flashing may have a variety of interpretations, depending upon the context of the signal. Uses of flashing include the following:

- To call an operator to the line upon completion of an operator-assisted call
- To signal a PBX that a special access or function code is to follow
- To place a call on hold

*Re-ring* is a signal used by an operator at the calling end to re-call an operator at the called end of an established connection.

*Re-call* is used by an operator at the called end of an established connection to re-call the originating operator.

*Receiver off-hook* is a tone combination sounded at 120 IPM. The receiver off-hook signal is used in conjunction with a recorded message to notify a subscriber that the receiver has remained off-hook for an abnormally long time.

## **Part I Trunking**

**TEST SIGNALS:** Installation and maintenance of trunks and circuits require several phases of testing, including pre-service, acceptance, ongoing routine testing, and trouble isolation. Testing is performed to ensure that the electrical characteristics of lines, trunks and transmission paths are adequate to meet stated objectives, such as the transmission of speech.

Test signals of a specified frequency and power level are generated and transmitted from one point in a network and received and measured at another point. The measured power and frequency parameters at the receiving end characterize the performance of the transmission path over which the test signals were carried.

Test signal generation and measurement are covered in detail in Part II of this book.

### **OUTPULSING METHODS USED IN ADDRESS SIGNALING**

In order for a transmission pathway to be set up between a calling and called subscriber, the address of the called subscriber generally must be transmitted over several links. When the signaling is accomplished over the voice transmission path (as opposed to Common Channel Interoffice Signaling), the address information proceeds from the calling to called subscriber in a link-by-link sequential manner. This may be accomplished through two types of operations.

In *non-senderized* operation, the calling subscriber must know the route that the call is to take through the network. Upon receiving dial tone from the first switch, the caller supplies an access code which tells the first switch the address of the second switch in the sequence. The call may proceed over several links, with the caller supplying the address of each successive switch. Once the last, or terminating, switch is reached, the caller provides the extension digits or telephone number of the called party.

In *senderized* operation, the first switching system receives complete address information from the calling subscriber over the line connecting the subscriber's set to the switch. From that point on, successive switches send the address information over successive links, based upon the routing structure of the network and the information contained in the address. Each switch must contain routing information in senderized operation.

### **A TYPICAL TELEPHONE CALL**

To illustrate the use of signals and the sequence in which they occur, let's follow a typical telephone call from beginning to end.

The phases of a typical call are:

- Origination
- Routing
- Answer Supervision
- Disconnect
- Clean-up

Figure 1-11 illustrates the network over which the call will be transmitted.

The calling party is served by Switch A, and the called party is served by Switch B in a nearby town. They are connected to their serving switches by line circuits, or local loops. Switch A is directly connected to Switch B by trunk group AB and indirectly connected through Switch C via trunk groups AC and CB. The called party's telephone number is 398-7324.

**Call origination** begins when the calling party lifts the receiver, which changes the state of his or her telephone set from *on-hook* to *off-hook* and seizes the line. This change of state is recognized by Switch A as a request for service. Switch A then connects the appropriate equipment to the calling subscriber's line and, once this has been accomplished, returns *dial tone* to the calling party, indicating that the switch is prepared to receive the called party's address.

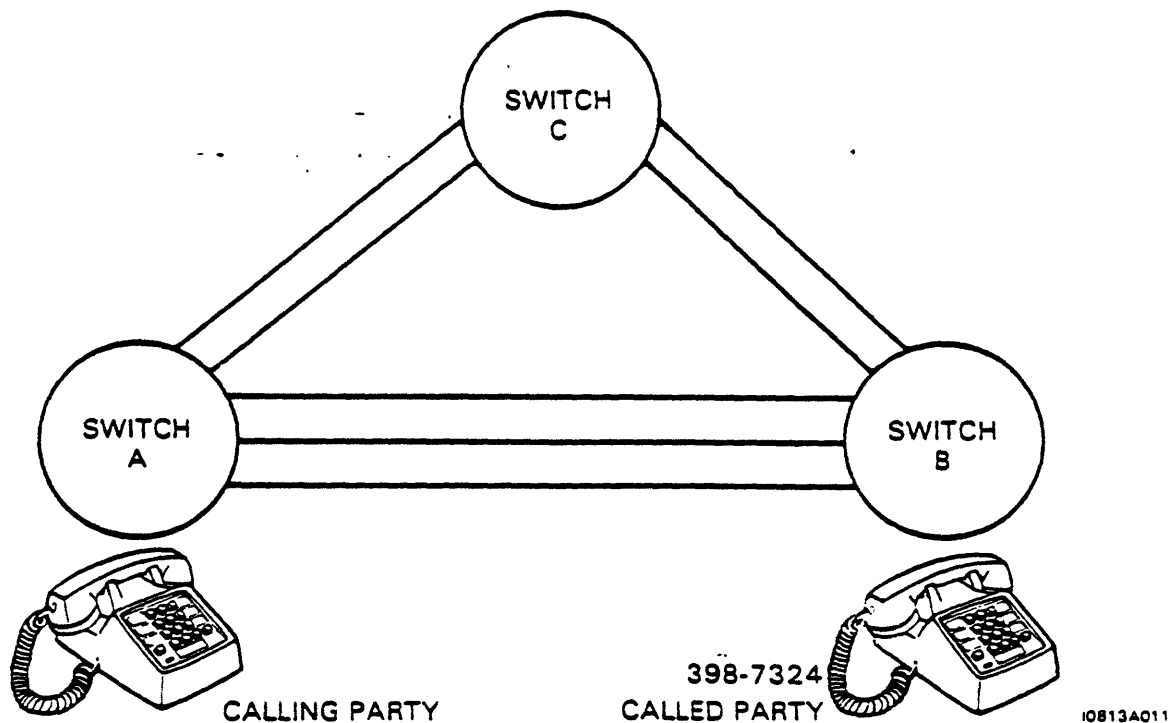


Figure 1-11. A Typical Telephone Call

The calling party sends the address digits, 398-7324, to the switch by dialing his or her rotary-dial telephone. As the dial returns after each digit is dialed, the dial mechanism momentarily changes the state of the telephone set from off-hook to on-hook a number of times equal to the dialed digit. The on-hook intervals are called *dial pulses*.

Switch A ceases to send dial tone once the first digit has been received. It then collects the digits as they are dialed and "remembers," or stores them, for later use. Call origination is complete once all digits have been dialed, received and stored.

Switch A then proceeds with the second phase of call processing, which is **call routing**. The switch consults an internal routing guide to determine which trunks may be used to connect with Switch B. The guide indicates that the direct trunk group, AB, is the *first route choice* and attempts to find an *idle* circuit in the trunk group. If an idle circuit is found in trunk group AB, Switch A will seize the idle circuit and proceed with call processing.

In this case, no circuit is available in trunk group AB. The switch again consults the routing guide and finds that an *alternate route choice* is the trunk group from Switch A to Switch C. Upon testing the circuits in this group, Switch A finds one idle and seizes the circuit. (In this example, trunk group AC is the *final route choice*. Had no circuits been available, the switch would have sent a *reorder* signal to the calling party.)

The seizure or off-hook condition on the selected circuit is recognized by Switch C and, once the appropriate equipment has been attached to the circuit, Switch C signals Switch A that it is prepared to receive address information. Upon receiving the address information from Switch A, Switch C consults its route translator, determines that the appropriate trunk over which to route the call is the direct route from Switch C to Switch B, and seizes an idle circuit within that trunk. Upon recognizing the seizure, Switch B requests and receives the required address information from Switch C.

## **Part I Trunking**

Switch B recognizes that the number is in the switch and tests the called line. Finding the line idle, it applies a *ringing signal* toward the called party's telephone, *alerting* the called party of an incoming call. A separate *audible ringing* tone is transmitted toward the calling party to indicate that the call is progressing.

When the called party answers, Switch B removes the ringing signal and the audible ringing tone toward the calling party. An off-hook signal is transmitted to Switch C and then relayed to Switch A. Detection of the off-hook or **answer supervision** signal at Switch A indicates that the called party has answered, and a record is made for accounting purposes.

The call has been completed; business can be conducted.

Once the conversation has ended, either party may initiate a **disconnect** by replacing the receiver in the handset, or going on-hook.

Once a disconnect is received by Switch A or Switch B (the last phase of the call), **call clean-up** takes place. Upon receiving the appropriate disconnect signals, Switches A, B and C restore the trunks involved in the call to their idle condition. (The call disconnect is recorded at Switch A for accounting purposes.)

## **CHAPTER 1 REVIEW QUESTIONS**

1. What is the advantage of a star network over a mesh network?
2. What is the difference between a line circuit and a trunk circuit?
3. Give four reasons for the development of private networks.
4. What is the purpose of CO trunks?
5. How does an FX trunk work?
6. Who pays for inward WATS calls?
7. What type of trunk is used to connect PBXs?
8. What is the advantage of DID trunks?
9. What is the purpose of OPX lines?
10. Briefly define signaling.
11. What are the two supervisory conditions?
12. Define percent break.
13. How many DTMF tones are available?
14. What is the difference between conventional signaling and CCIS?
15. List five information signals.
16. What are the two outpulsing methods used in address signaling?
17. What are the phases of a telephone call?
18. What happens when a switch runs out of route choices?



## CHAPTER 2 OVERVIEW OF CIRCUIT OPERATION

### INTRODUCTION

In Chapter 1 we saw how line circuits, trunk circuits and switching systems are assembled to form telephone networks. Various types of trunks and circuits were discussed according to the type of service each provides. The types of signaling that are required, as well as the functional operation of each, were introduced.

This chapter introduces the types of *transmission facilities* that are used in telephone networks. *Signaling systems* used over these facilities are also introduced. The concept of *interfaces* between different transmission facilities and signaling systems is presented and illustrated with examples of commonly encountered interfaces: the *demarc*, *4-wire termination set* and *E&M signaling*.

A walk-through of an end-to-end connection is presented in order to illustrate types of facilities and signaling systems and the interfaces between differing types.

*Specific signaling formats*, which will be detailed in later chapters, are also introduced.

### INTRODUCTION TO TRANSMISSION FACILITIES

Although transmission will be covered in detail in Part II of this book, an introduction to the terminology used in transmission will assist the reader in understanding signaling operation.

In this chapter we will be concerned with two major categories of transmission facilities: *metallic* and *analog carrier*.

**METALLIC TRANSMISSION FACILITIES:** Metallic facilities consist of pairs of wires that are capable of carrying *direct current* (dc) and *tone*. There are two types of metallic facility: a 2-wire metallic facility and a 4-wire metallic facility.

A *2-wire metallic facility* consists of a single pair of wires that carry signals and voice in both directions. The two conductors in a 2-wire facility are designated Tip (T) and Ring (R), nomenclature that derives historically from the parts of a standard telephone plug to which they were connected in the days of manual switchboards (see Figure 2-1).

A block diagram of an end-to-end connection using a 2-wire facility is shown in Figure 2-2. The arrows indicate that transmission in both directions is carried over the single pair of wires.

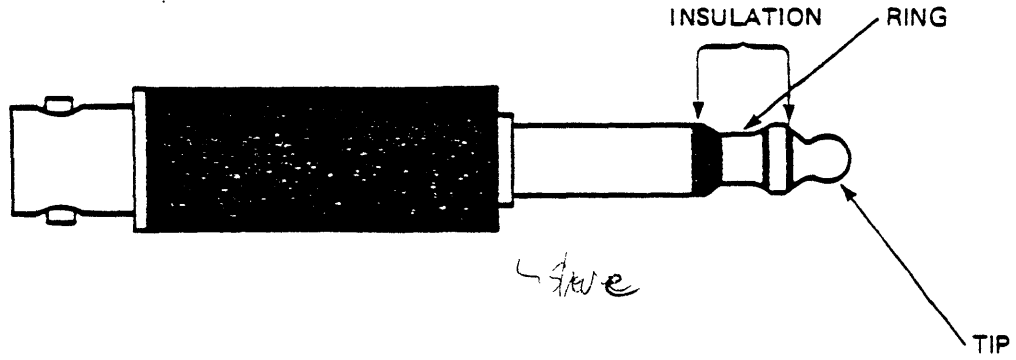
The 2-wire facility functions well over short distances and is generally used between a subscriber's telephone set and a CO or PBX. The 2-wire facility may also be used between a PBX and a CO or may be used for short connections between COs.

A *4-wire metallic facility* requires two pairs of wires, each pair carrying information in only one direction. The four wires are called T, R, T1 and R1. Figure 2-3 illustrates a 4-wire connection between two PBXs.

By convention, the T and R leads are used to transmit from the switching equipment toward the facility; T1 and R1 are used to receive from the facility.

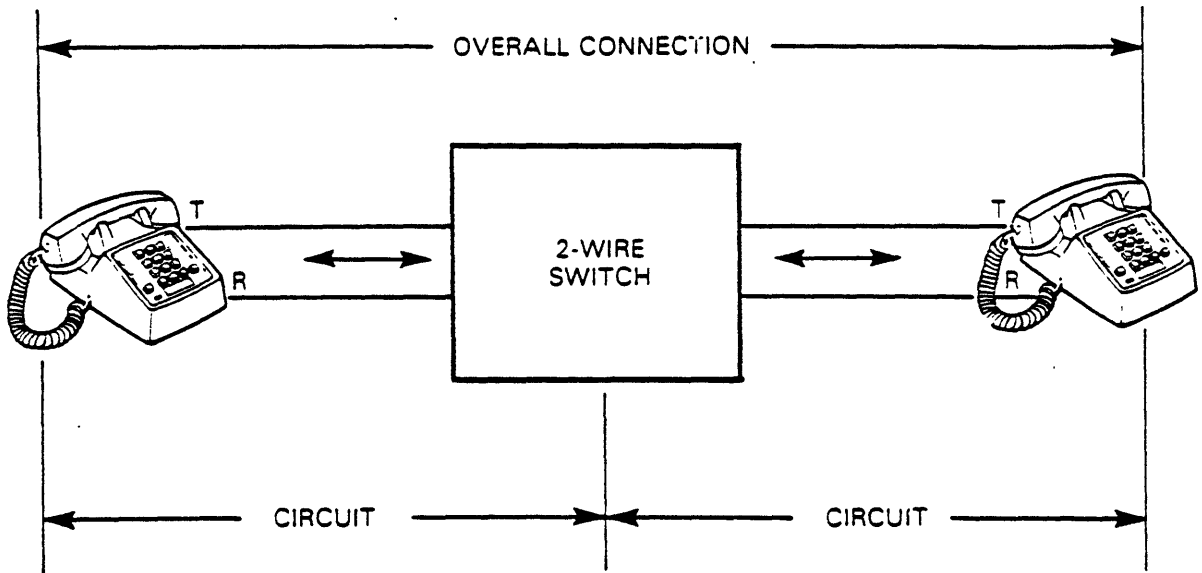
The 4-wire facility is more expensive than the 2-wire; however, it has the advantage of extending the range of metallic facility transmission. Another advantage of 4-wire transmission is that it may be interfaced directly with long distance transmission facilities, which separate the transmission directions just as 4-wire transmission does.

**Part I**  
**Trunking**



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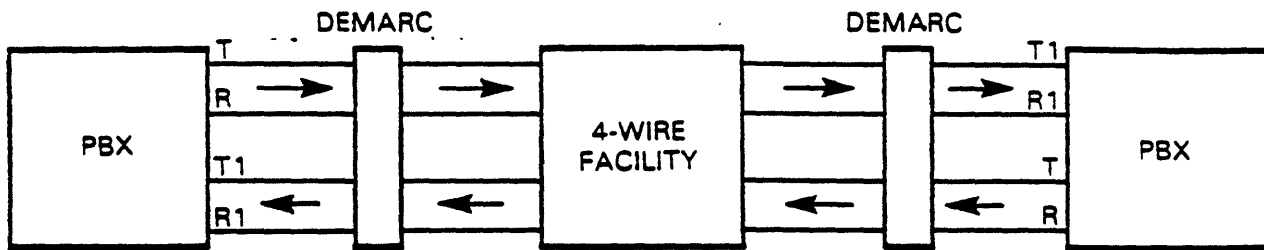
**Figure 2-1. Standard Telephone Plug**



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**Figure 2-2. An End-to-End 2-Wire Connection**





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Figure 2-3. A 4-Wire Connection

**ANALOG CARRIER TRANSMISSION FACILITIES:** Analog carrier facilities are used for long-haul transmission. *Transmission media* that are used in analog carrier facilities are:

- Paired cable
- Coaxial cable
- Radio (terrestrial and satellite)
- Waveform
- Optical fibers

With the exception of paired cable, the transmission media used in analog carrier facilities do not have a continuous metallic path and, therefore, are not capable of carrying direct current. All signaling and transmission are accomplished by *modulating* a high-frequency electromagnetic wave (the *carrier*) with the analog waveform that carries the information to be transmitted. A major economic advantage of carrier facilities is their capability of carrying many simultaneous conversations over a single transmission path. This is achieved through a technique called *multiplexing*.

In a multiplexed facility, each conversation is assigned two specific frequencies (one for each direction of transmission). Each frequency is modulated with the analog signaling or speech waveform, and all of the frequencies (or *channels*) are transmitted over the same path (e.g., the line of sight path between two microwave towers).

The important thing to remember here is that direct current cannot be used for signaling on analog carrier facilities.

## **INTRODUCTION TO SIGNALING SYSTEMS AND FORMATS**

Several types of transmission facilities may be encountered in an end-to-end connection. This has resulted in two types of *signaling systems*—*dc* and *tone*. Within both systems, there is more than one basic methodology for conveying signals, and these methodologies are called *signaling formats*. Each signaling format may be operated in several ways. As mentioned earlier, dc signaling may only be used over metallic facilities and is, therefore, *facility dependent*.

Table 2-1 summarizes the signaling systems and formats that we will be concerned with in this book.

**Table 2-1. Signaling Summary**

<b>SIGNALING SYSTEMS</b>		<b>SIGNALING FORMATS</b>
<b>DC SIGNALING SYSTEMS</b>	<b>TONE SIGNALING SYSTEMS</b>	
Loop Signaling E&M Signaling Duplex Signaling (DX)	In-Band Tone Signaling Out-of-Band Signaling	Loop Start Ground Start Delay Dial Wink Start Direct Inward Dialing

### **DC SIGNALING SYSTEMS**

The types of dc signaling systems used over metallic transmission facilities are:

- Loop Signaling
- E&M Signaling
- Duplex Signaling

**LOOP SIGNALING:** *Loop signaling* uses dc signals that originate or terminate in station sets or switching machines. The dc signals are used for supervisory signaling and also for address signaling when dial pulse is used. In addition to dc signals, loop signaling employs a 20 (or possibly 30) Hz signal, applied toward the subscriber's telephone set in order to ring the bell (alerting signal).

Tone signaling is used in conjunction with loop signaling to provide signals in categories other than supervisory, alerting and address. (Tone is used for address signaling when DTMF is used.)

Loop signaling directly operates a relay or other sensing device to indicate seizure, dial pulse and disconnect. We will be concerned with two types of loop signaling, *Loop Start* and *Ground Start*, which use different methods to indicate changes of state between on-hook and off-hook.

**E&M SIGNALING:** E&M signaling, though often considered a signaling format or type, is considered in this book as a *signaling interface* and is discussed in detail in the Interface section on pages 2-6 through 2-9. The E&M signaling interface uses dc signaling in order to provide an interface between different signaling systems.

**DUPLEX SIGNALING:** Duplex (DX) signaling uses the metallic pair(s) of the talking path to extend the dc signaling range. DX signaling may be used over either 2-wire or 4-wire metallic facilities, and is only used in conjunction with the E&M signaling interface to extend the E&M signaling range.

## **TONE SIGNALING SYSTEMS**

Tone signaling systems are used over both metallic and analog carrier transmission facilities. These systems are:

- In-Band Tone Signaling
- Out-of-Band Tone Signaling

**IN-BAND TONE SIGNALING:** In-band tone signals are signals that fall within the *voice frequency band*, which is defined as approximately 200 to 3400 Hz.

*DTMF* and *MF* in-band tones, used for address signaling and interoffice address and control signaling, respectively, were discussed in Chapter 1.

*Single Frequency (SF)* signaling, consisting of a 2600 Hz signal, is used to transmit supervisory and address signaling over transmission facilities. Signaling is accomplished by changes of state from presence of tone to absence of tone, and vice-versa.

**OUT-OF-BAND TONE SIGNALING:** Out-of-band tone signaling uses tones with a frequency above the voice frequency band to convey supervisory and address signaling.

## **INTERFACES**

As we have seen, a variety of transmission facilities and signaling types is used in telephone networks. In many cases, several types of transmission facilities are interconnected to provide an end-to-end path between subscribers, and signaling will be of several different types along this path.

In general, an interface is the point, piece of equipment, or technique that is used to connect two unlike pieces of equipment or methods of operation.

A basic function of an interface is to provide a method by which the responsibilities and functions on either side of the interface may be made, at least to a large extent, independent of one another. If this is accomplished, changes on one side of the interface will not require changes on the other side. Through the use of standardized interfaces and standardized interface specifications, changes to interfaces in telephone networks are often straightforward and relatively inexpensive.

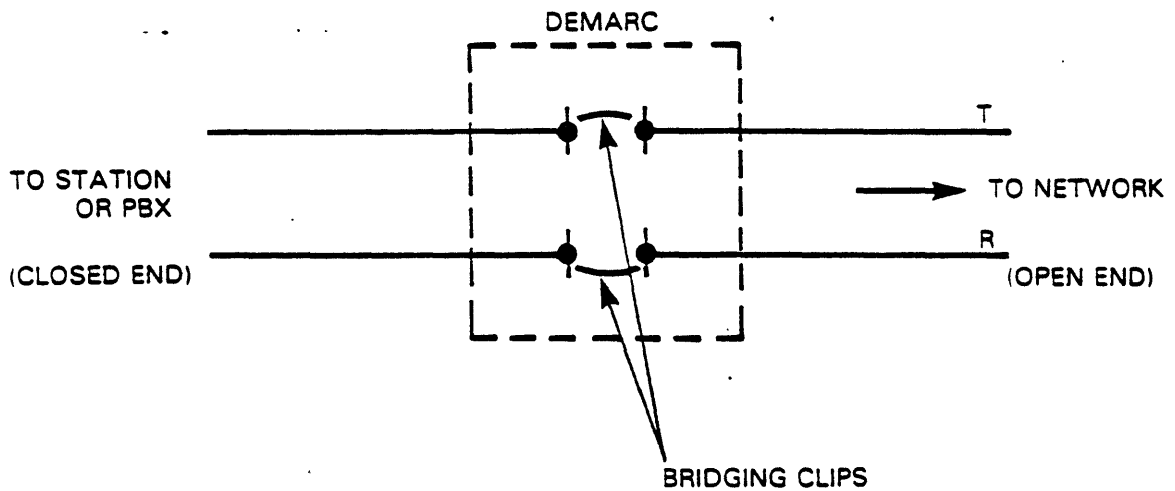
A second important function of an interface is the provision of a point where two interconnected systems can be tested independently. If the systems or pieces of equipment on either side of the interface perform according to specification when they are tested independently, it can be reasonably assumed that they will function properly when interconnected.

To illustrate the interface concept, let's look at three interfaces that are commonly encountered in telephone networks.

**DEMARCATION POINT:** An interface that is straightforward in concept is a demarcation point, commonly referred to as a *demarc*. A 2-wire demarc is illustrated in Figure 2-4.

With the bridging clip removed, signaling in the 2-wire circuit may be tested for correct operation toward the network (open end) and toward the customer equipment (closed end). (Operation of loop start and ground start signaling types is described in later chapters in terms of "looking" toward the open and closed ends from the demarc.)

## Part I Trunking



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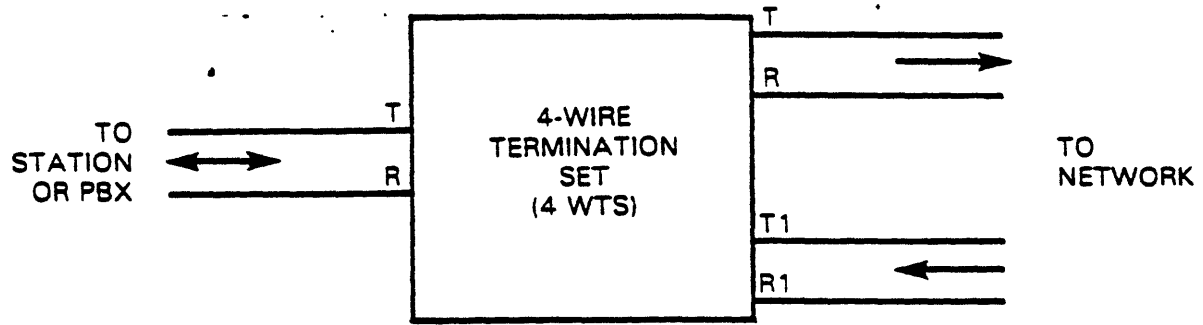
Figure 2-4. A 2-Wire Demarc .

**4-WIRE TERMINATION SET:** As discussed earlier, 2-wire transmission (and its associated loop signaling) is commonly used for subscriber lines. The reason for this is primarily economic, as 2-wire circuits require half the metallic conductor needed for 4-wire circuits. However, 2-wire transmission has several limitations which preclude its wider use. It is limited to relatively short distances (usually stated in terms of the resistance of the loop) and requires amplification beyond a certain distance. Amplification in order to extend the range of 2-wire facilities can be expensive due to the additional equipment that is required and the fact that each loop must be individually engineered and maintained. Many switches (including PBXs) are designed to interface with 4-wire circuits, and all analog carrier channels require separate transmit and receive paths.

Conversion between 2-wire and 4-wire transmission is frequently required. The 4-wire termination set (also commonly referred to as 4 WTS or a *hybrid circuit*) provides the interface between 2-wire circuits and 4-wire circuits. A block diagram of a 4-wire termination set is shown in Figure 2-5

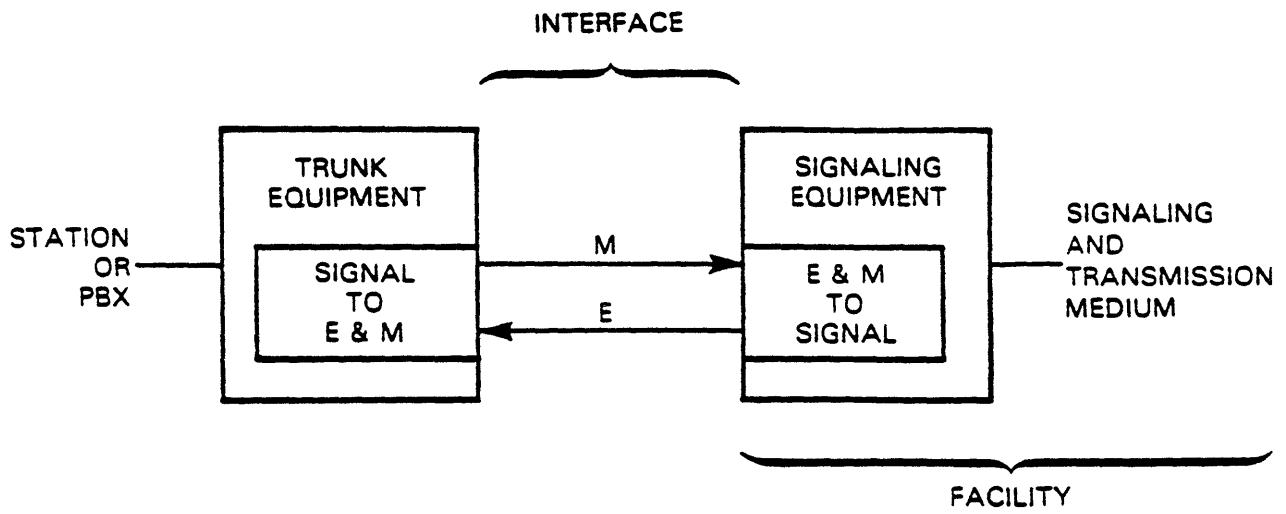
The bidirectional transmission in the 2-wire circuit is split by the hybrid. By convention, the transmit direction 4 WTS is as shown and is carried on the T and R leads. The receive direction is carried on T1 and R1.

**E&M SIGNALING INTERFACE:** Another important interface is between the signaling equipment in one switching system and the transmission equipment that carries the signals to a distant switching system. One of the most common interfaces in this category is the E&M lead interface. The E&M interface is standard and widely used, and conversion units are available to convert most signaling types to and from E&M signaling. The E&M interface is designed to transmit and receive two states, on-hook and off-hook. The block diagram in Figure 2-6 illustrates the direction conventions that are standard for the E&M interface used to connect signaling systems together



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**Figure 2-5. 4-Wire Termination Set**



10813A017

**Figure 2-6. E&M Signaling Interface**

## Part I Trunking

The on-hook or off-hook state on the drop side is transmitted *from* the trunk equipment (usually co-located with the switching equipment) to the signaling equipment part of the facility on the M lead. The appropriate conversion is performed by the signaling equipment (e.g., E&M to SF) and the state is then sent to the distant office over the transmission medium. Incoming signals are similarly converted by the signaling equipment and are received by the trunk equipment on the E lead.

There are three types of E&M signaling: Types I, II and III. We will discuss Types I and II. The original E&M signaling (Type I) is illustrated in Figure 2-7a. Only one lead is used for each direction of transmission. This means that current flowing between trunk equipment and signaling equipment must return over a common ground path. While Type I E&M signaling circuits operate well in electro-mechanical switching systems, they may not function satisfactorily in electronic systems, primarily due to noise considerations. Consequently, the Type II interface was designed for use in electronic systems.

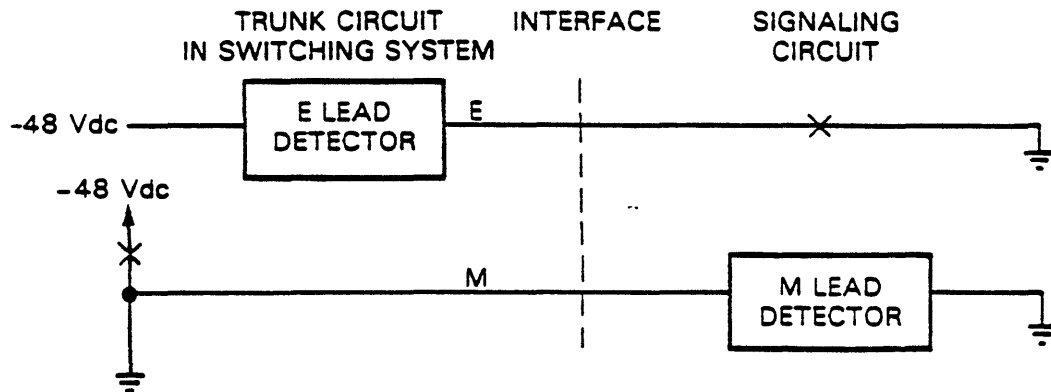
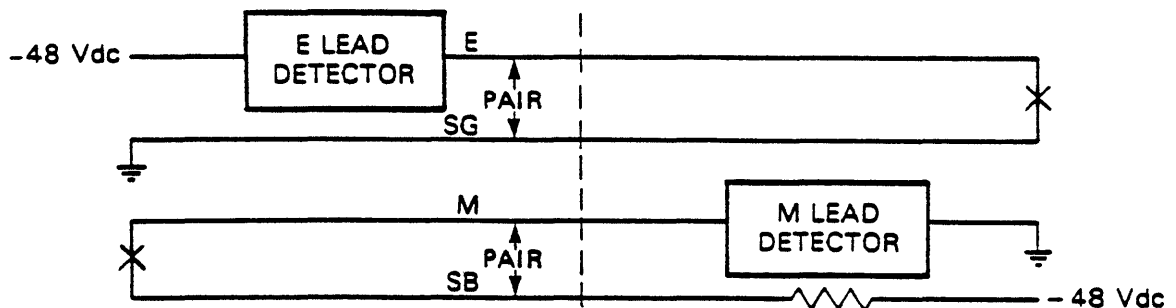


Figure 2-7a. E&M Signaling—Type I

The Type II interface, illustrated in Figure 2-7b, is fully looped; that is, both E&M leads have their own ground return path.



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Figure 2-7b. E&M Signaling—Type II

The conventions used to represent on-hook and off-hook states for E&M Types I and II are summarized in Table 2-2.

The details of E&M signaling operation and E&M conversion to and from various signaling types are given in subsequent chapters. The important thing to remember here is that E&M signaling is a *signaling interface* and should not be considered as a *signaling type*.

**Table 2-2. E&M Lead States**

	STATE	FROM CO TO FACILITY (M Lead)	FROM FACILITY TO CO (E Lead)
TYPE I	On-Hook	Ground	Open
	Off-Hook	- 48 Vdc	Ground
TYPE II	On-Hook	Contact open	Contact open
	Off-Hook	Contact closed	Contact closed

### **END-TO-END CHANGE OF STATE**

To illustrate the signaling and transmission conversions and interfaces involved in a typical end-to-end connection, let's trace the change from on-hook to off-hook through the OPX connection represented by the block diagram in Figure 2-8.

The transmit and receive directions over the leads are indicated in the figure. To clarify direction conventions, it is useful to introduce the concept of line and drop. At any point in an end-to-end connection, the line side is toward the transmission facility, while the drop side is toward the equipment that is used for call origination and termination. The line and drop directions are indicated for selected points in Figure 2-8.

Signaling from the drop side to the line side is transmitted by the M lead; signaling from the line side to the drop side is transmitted by the E lead. Transmission from the drop side to the line side is carried by T and R; transmission from the line side to the drop side is carried by T1 and R1. This is summarized in Table 2-3.

**Table 2-3. Line and Drop Conventions**

	FROM DROP TO LINE	FROM LINE TO DROP
TRANSMISSION	T and R	T1 and R1
SIGNALING	M	E

When the subscriber at A lifts the receiver, the change of state from on-hook to off-hook is signaled to the demarc via loop signaling. No signaling or transmission conversion is necessary at the demarc, and the off-hook state is propagated to Central Office A.

At CO A, two conversions take place. The 2-wire transmission is converted to 4-wire via the 4 WTS interface, and loop signaling is converted to E&M signaling by the loop to E&M conversion unit. The off-hook state is signaled over the M lead to the signaling equipment (contained within the Analog Facility Terminal (AFT)), causing it to turn 2600 Hz SF off.

The change in state is transmitted over the analog carrier facility to the AFT in Central Office B. No signaling or transmission conversion is required at the incoming AFT in CO B, as the facility remains 4-wire.

## Part I Trunking

The office equipment that connects the two facilities in CO B is taken to be the "drop side." (This makes sense, as local lines could actually "drop" from the facility at this point.) Therefore, analog transmission and SF signaling come into the demarc from the AFT on T1 and R1 (from line to drop) and leave the demarc on T and R (from drop to line).

At CO C, SF signaling is converted to E&M signaling, and the change of state is signaled from the AFT to the MFT on the E lead (from line to drop).

The MFT on the drop side of CO C converts E&M signaling to DX signaling, which is one of the signaling formats used in dc signaling. DX may be thought of as extending the range of E&M signaling, and the off-hook signal received by the MFT is extended to the customer's interface over the wire pairs using a convention known as *simplexing*.

The customer's interface converts DX signaling to E&M signaling, and the off-hook state is transmitted to the PBX over the E lead. In this case the PBX then returns dial tone to the originator at A.

This example illustrates that E&M signaling is a signaling interface rather than a signaling format or type. In CO A, E&M was used as an interface between the incoming loop signaling and the outgoing SF signaling; in CO C, E&M signaling provided the interface between SF and DX signaling.

While some other form of interface could have been used, E&M operation provides a standard and well understood interface, which simplifies conversion between signaling types and lends standardization to supervisory and address signaling within the switches.

### SPECIFIC SIGNALING FORMATS

The purpose of this section is to introduce the specific signaling formats that will be discussed in detail in the following chapters. These signaling formats, which may use different signaling systems, are:

- Loop Start
- Ground Start
- Delay Dialing
- Wink Start
- Direct Inward Dialing

**LOOP START:** Loop start signaling uses the loop signaling format. On-hook and off-hook states are represented by the absence or presence of current in the loop.

Loop start signaling is the simplest type of signaling, and the equipment needed for loop start is relatively inexpensive. Loop start is used for signaling over subscriber line circuits, or loops—that is, over circuits between station sets and switches (either CO or PBX).

Loop start is sometimes used on CO trunks; however, there are significant problems associated with loop start operation between switches.

One problem is the possibility of simultaneous seizure of both ends of the trunk (i.e., by the PBX and the CO). This condition, called *glare*, is not guarded against in loop start operation.

Another problem is that there is no requirement for the CO to inform the PBX of a distant-end disconnect on loop start facilities. In order to allow disconnect recognition in either direction, CO trunk signaling should be ground start.

**GROUND START:** As with loop start, ground start operation uses the loop signaling format. Ground start is used for CO trunks. The glare problem is minimized with ground start, and ground start allows the PBX to recognize a distant-end disconnect.





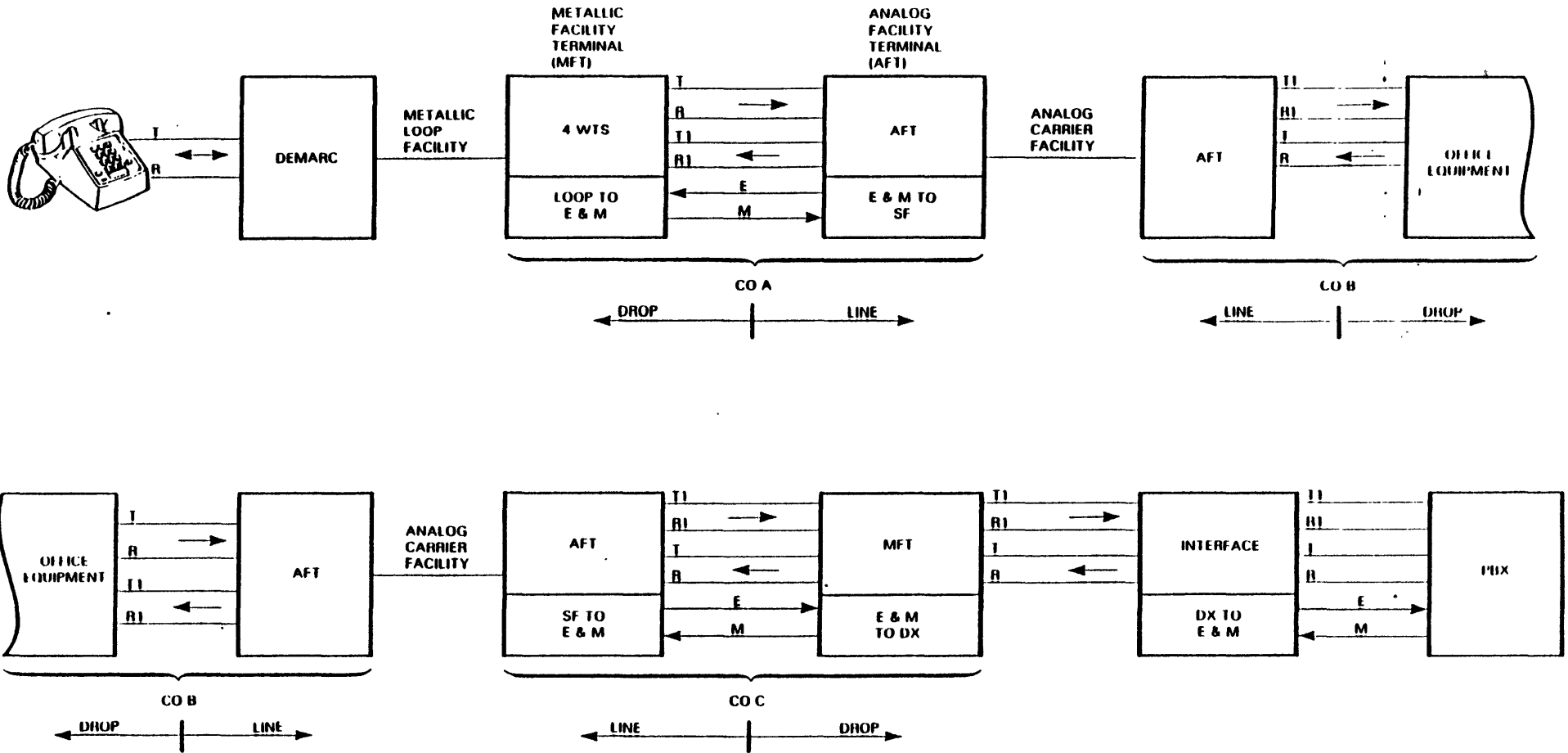
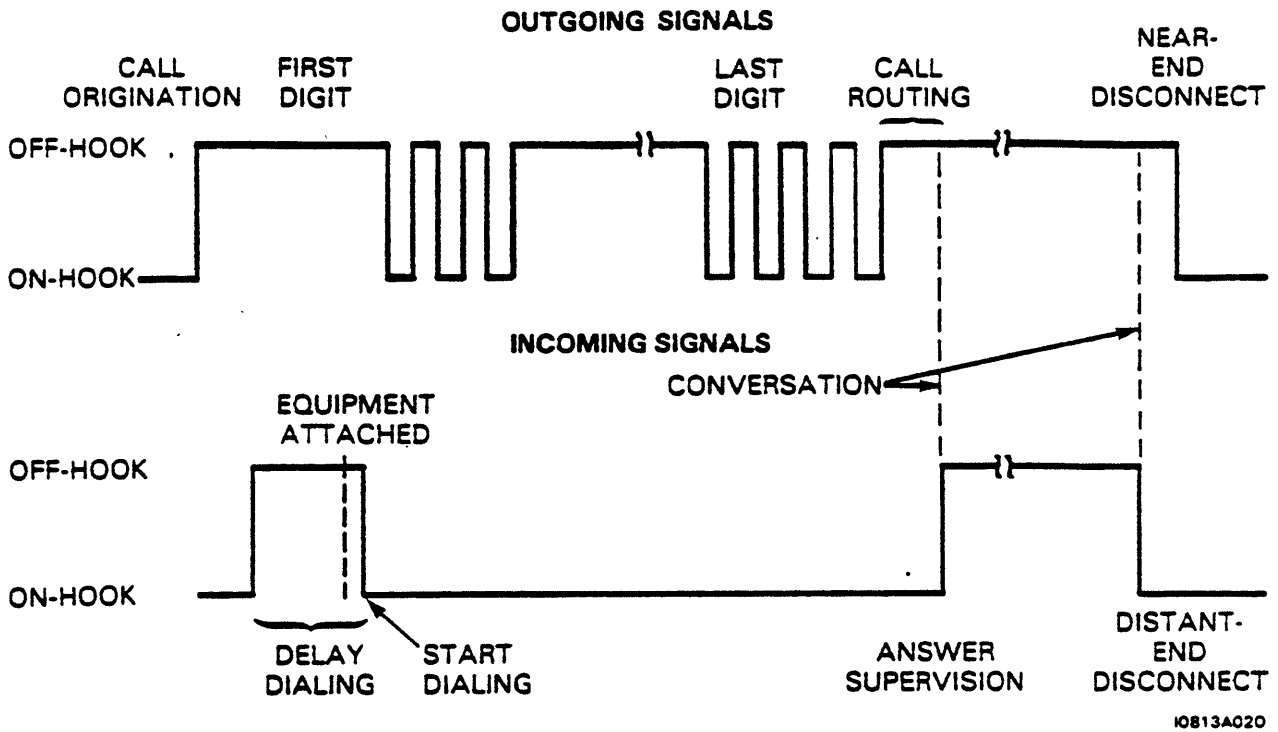


Figure 28 End-to-End Change of State

**DELAY DIALING:** Delay dialing operation is used on interoffice trunks. When the called switch recognizes a trunk seizure from the calling switch, it must locate the appropriate equipment and attach it to the circuit before it can receive address information. The called office signals off-hook to the calling office (delay dialing signal) until it is prepared to accept the address digits; at that point, it sends the off-hook to on-hook transition (start dialing signal). Figure 2-9a illustrates the timing of delay dialing signaling.



**Figure 2-9a. Delay Dialing Timing**

**Part I  
Trunking**

**WINK START:** Wink start operation is used for interoffice signaling. On receipt of a trunk seizure from the calling office, the called office does not immediately return an off-hook signal to the calling office. The idle-condition, on-hook signal is maintained until the appropriate equipment is attached, at which time the called office changes the on-hook condition to off-hook. The off-hook duration, or "wink" duration, is between 140 and 200 milliseconds. Upon receiving the wink, the calling office sends the address signals. Figure 2-9b illustrates the timing of wink start signaling.

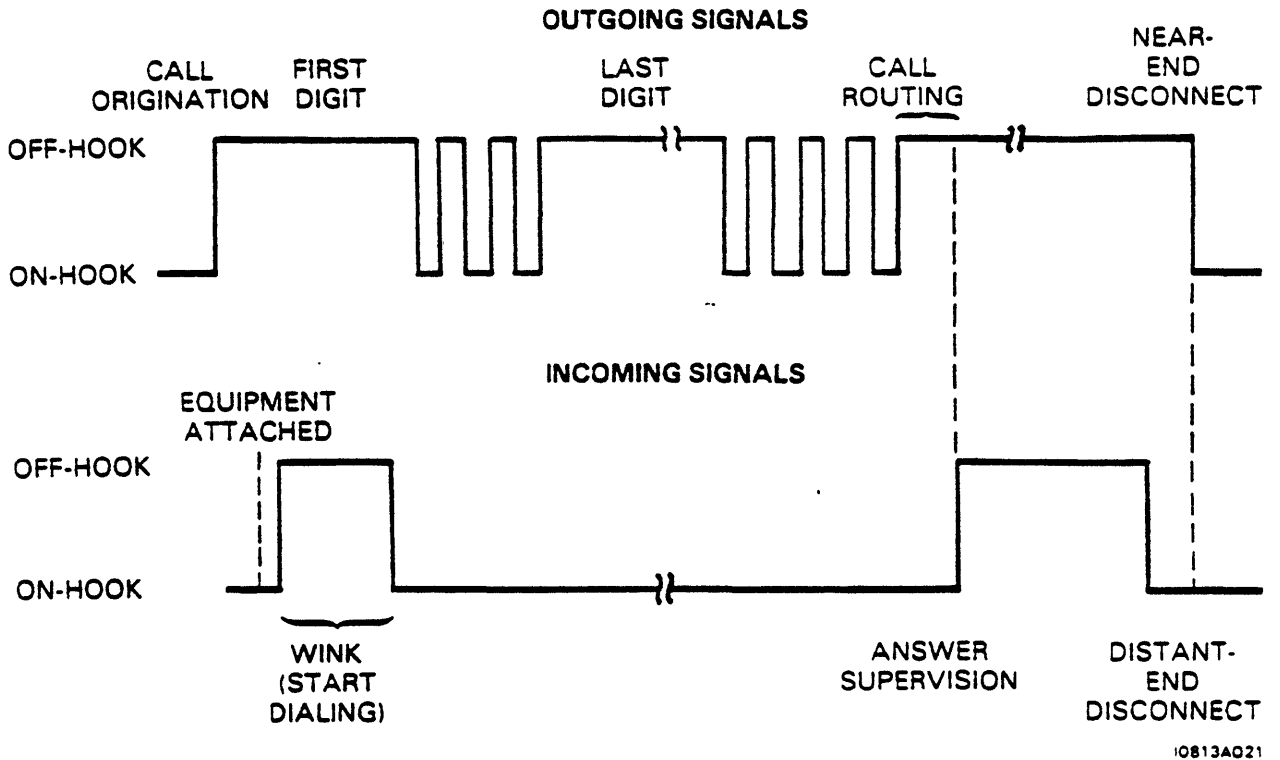


Figure 2-9b. Wink Start Timing

**DIRECT INWARD DIALING:** Direct Inward Dialing (DID) uses the loop signaling format over special CO trunks that are set up for DID. Upon receiving an incoming DID trunk seizure from the CO, the PBX attaches the appropriate digit collection equipment to the line prior to signaling off-hook to the CO. The CO transmits extension digits to the PBX, which then rings the appropriate station.

## **CHAPTER 2 REVIEW QUESTIONS**

1. What is the major difference between metallic and analog carrier facilities?
2. Give the transmission direction conventions for 4-wire circuits.
3. What are the two types of signaling systems?
4. What is the function of 4-wire termination sets?
5. Define glare.
6. What are two methods of controlled signaling?



## CHAPTER 3 LOOP START SIGNALING

### INTRODUCTION

Loop start signaling uses a loop signaling format. The major application of loop start signaling is with station circuits connecting a telephone with a switching machine (PBX or Central Office switch).

Loop start signaling may also be used to connect a PBX with a Central Office switch. This text is primarily concerned with signaling over shared circuits, or trunks; therefore, loop start will be discussed in terms of its application to CO trunk signaling.

It is important to recognize that loop start signaling over CO trunks has three major disadvantages when compared to ground start:

- Loop start is more susceptible to *glare*, or simultaneous seizure at opposite ends, than is ground start.
- A PBX not equipped with dial tone detection equipment has no positive way of knowing when a CO is prepared to accept address digits.
- Loop start does not provide positive signaling to the PBX of a distant-end disconnect.

These disadvantages will be discussed in this chapter and also in Chapter 4, Ground Start Signaling.

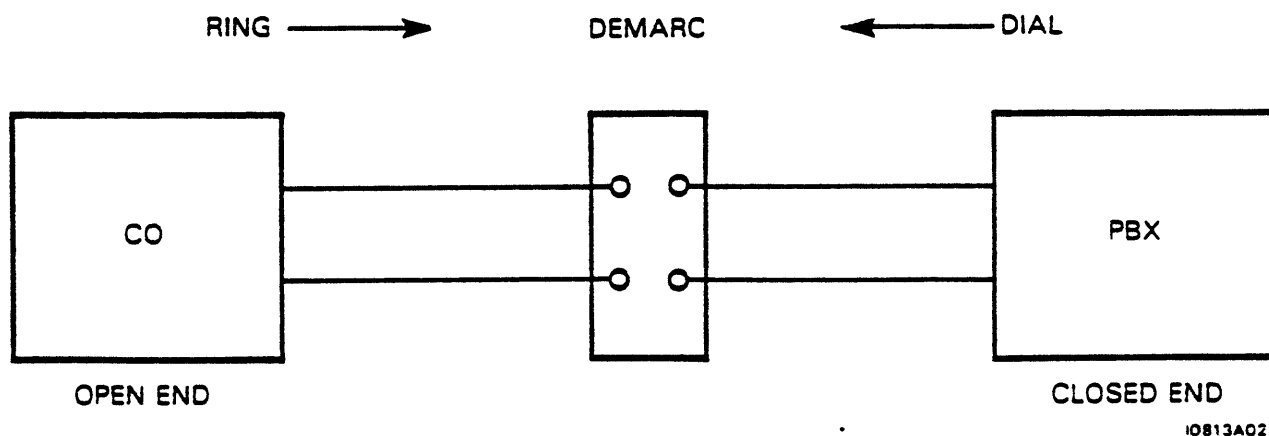
This chapter begins with an overview of loop start operation. The overview is followed by a detailed discussion of loop start signaling with reference to block diagrams.

The E&M to loop start interface is presented with reference to the signaling conversions that take place.

Appendix A contains a discussion of the overall operation of loop start circuitry and the overall operation of loop start to E&M interface circuitry. The discussions are conducted with reference to simplified circuit diagrams.

### LOOP START OVERVIEW

This section provides an overview of loop start signaling over a 2-wire line circuit between a PBX and a Central Office. Figure 3-1 illustrates the connection. As indicated in the figure, an incoming call to the PBX is initiated by ringing from the CO. The PBX dials outgoing calls. This is called a one-way ring, one-way dial operation.



**Figure 3-1. PBX/Central Office Connection**

## **Part I Trunking**

**IDLE CONDITION:** Looking toward the Central Office switch, the ring lead is attached to the battery and the tip to ground during the idle state. Both tip and ring are open looking toward the PBX.

**OUTGOING CALL:** An outgoing call is originated when the PBX applies a loop closure across the tip and ring leads, causing current to flow in the loop between the PBX and the Central Office. The CO senses the current flow and attaches a digit receiver to the circuit in preparation for receiving address information. Once the equipment is attached, the CO provides dial tone to the PBX.

Many PBXs do not have the capability of recognizing dial tone and, in senderized operation, simply wait a specified period of time between closing the loop and outpulsing address information. If the wait interval is too short, the PBX will outpulse before the CO is prepared to receive digits, resulting in an incomplete or incorrectly routed call. In non-senderized operation, dial tone is passed through the CO and is heard by the user.

In the senderized mode, once all the digits have been outpulsed and collected by the CO, an audio path is established between the call originator and the CO.

Any further call progress tones are transmitted directly to the originator as audible signals.

**INCOMING CALL:** An incoming call from the CO is originated when the CO applies an interrupted ring (2 seconds ring, 4 seconds rest) on the facility toward the PBX. Upon sensing the incoming ring, the PBX applies a loop closure across the tip and ring leads...

The CO detects the loop closure, disconnects the ringing supply, and establishes an audio path to the PBX.

The PBX may route the call to a specific location or to an operator for further routing. In certain operations, the PBX transmits dial tone to the calling party to inform the caller that the PBX is ready to receive further address information.

It should be noted that dial pulse address signaling cannot be transmitted to the PBX by the CO. This is because the CO has no way of opening and closing the loop (pulsing) under the conventional loop start signaling format. A CO line switch can only ring toward its drop side.

One way to obtain dial-in capability is to use DTMF, which requires that the originator be capable of sending DTMF digits. If incoming DTMF addressing is to be allowed, power losses over incoming connections may occur, causing DTMF levels that are too low to be detected by the PBX. This is particularly true of calls traveling a great distance over the public switched network, where the DTMF signals may have traversed several tandem links over a considerable distance.

During the time between the routing of the call to the facility by the CO and the ringing condition being sensed by the PBX, the circuit appears idle to the PBX and, therefore, may be seized for an outgoing call. Simultaneous (or nearly so) seizure of opposite ends of a facility is called *glare*.

During the 4-second rest period of the incoming ring, the circuit again can appear to be idle to the PBX. Seizure by the PBX during this period also results in a glare condition.

When glare occurs, it results in either misdirected calls or calls that are not completed.

The incidence of glare can be significantly reduced by using ground start operation with CO trunks.

**CALL DISCONNECT:** Either the calling or called party may disconnect the call by going on-hook. However, a distant-end disconnect signal is not required to be relayed to the PBX by the CO under the loop signaling format. (There is no change of state; ring remains connected to battery, and tip to ground.) Special equipment must be available at the CO to send a disconnect signal.

The only disconnect that the PBX can recognize is when the PBX user goes on-hook. The PBX recognizes the station circuit on-hook and opens the loop closure toward the CO, thereby disconnecting the call.



Loop start works very well for line circuits that connect telephones to switches, a human has no problem recognizing a distant-end disconnect.

In order to allow disconnect recognition from either direction, ground start operation should be used with CO trunks.

### **DETAILED OPERATION OF LOOP START SIGNALING**

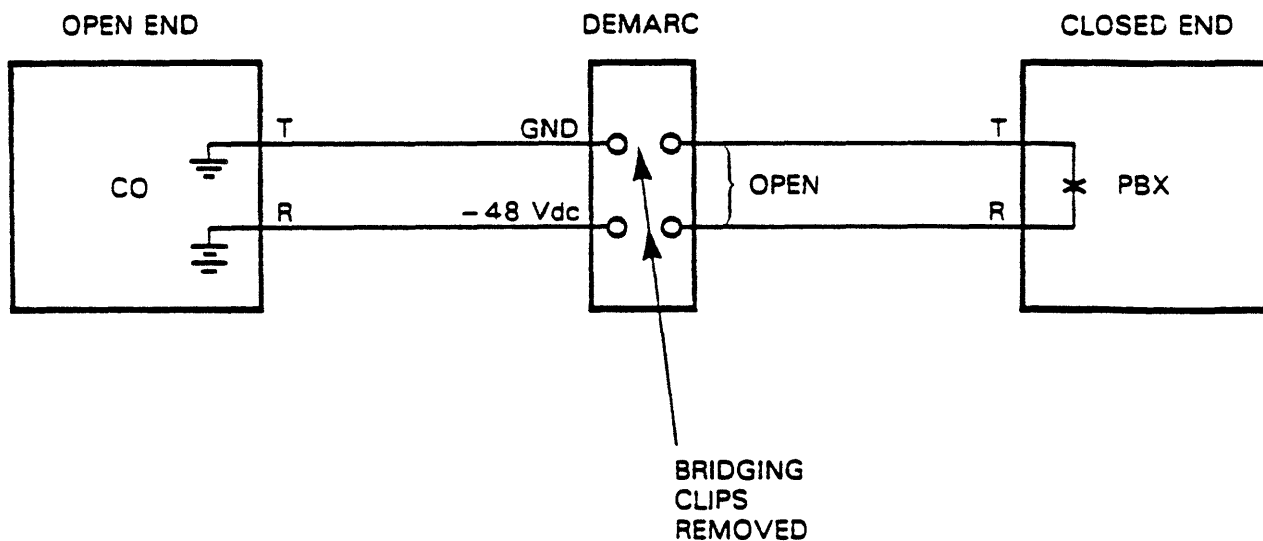
The signaling states of loop start operation will be discussed with reference to two viewpoints: from the demarc looking toward the open end (the CO) and from the demarc looking toward the closed end (the PBX).

Although this section is not intended as a guide to circuit testing, the signaling operations are described in much the same manner as they would be observed during testing.

**IDLE CONDITION:** The idle condition is represented in Figure 3-2. The bridging clips have been removed to isolate the open and closed ends.

Looking toward the PBX (closed end), an open condition is observed between the tip and ring leads at the demarc.

Looking toward the CO (open end) from the demarc, ground is observed on the tip lead and -48 Vdc is observed on the ring lead.

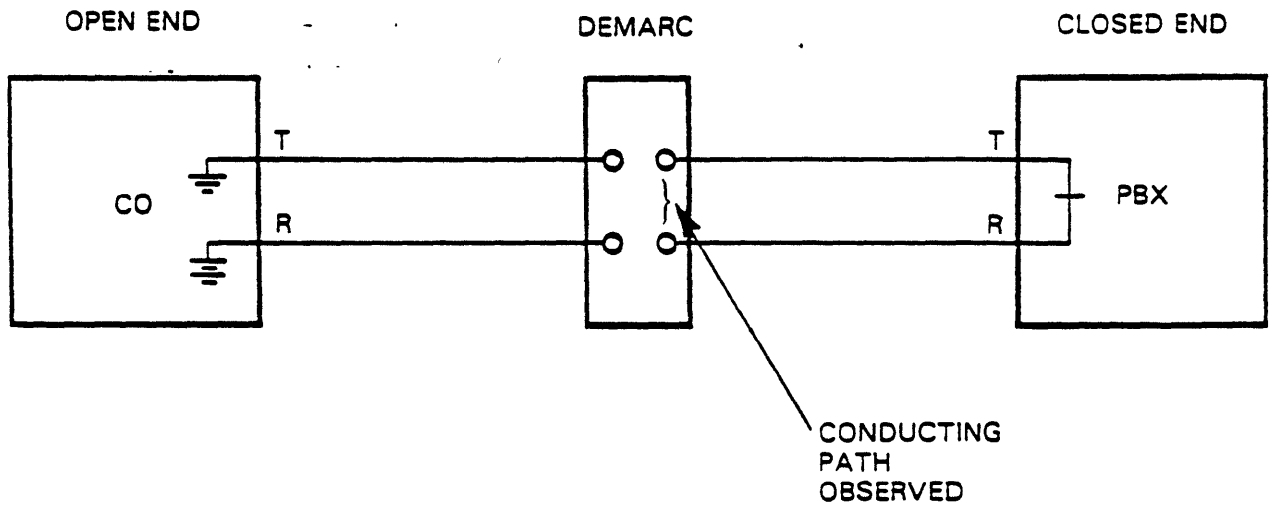


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**Figure 3-2. Idle Condition**

## Part I Trunking

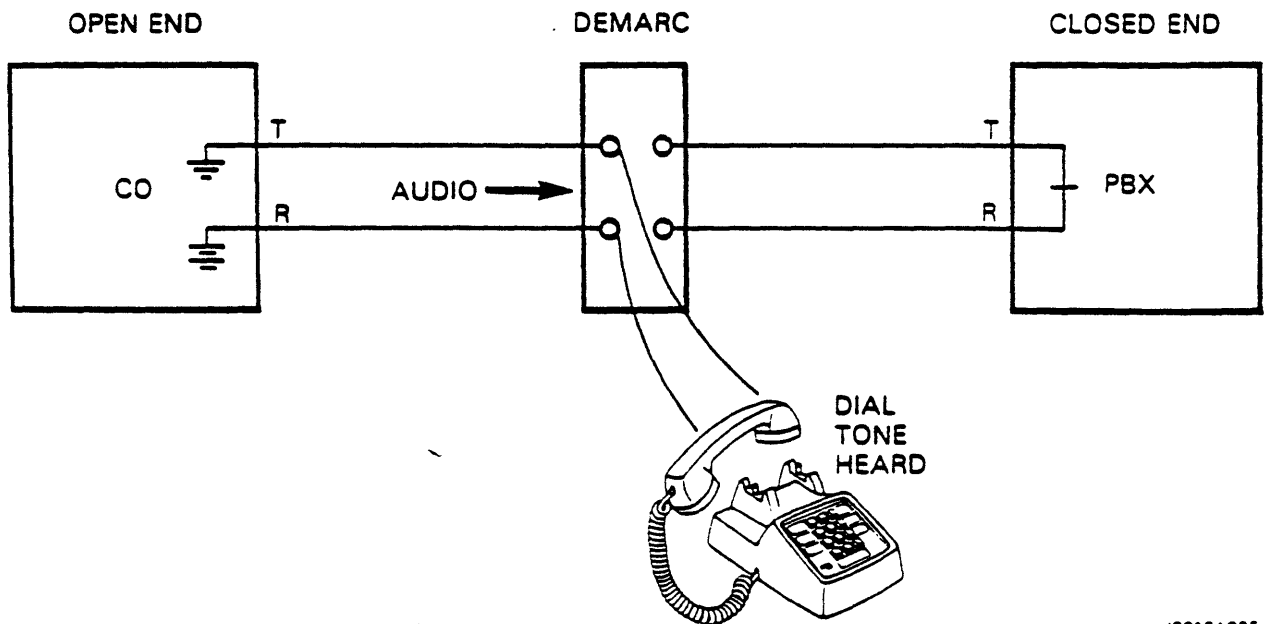
**OUTGOING CALL:** An outgoing call is originated by loop closure in the PBX (Figure 3-3). With the bridging clips removed, the closure is observed as a conducting path (low resistance) between the tip and ring at the demarc looking toward the PBX.



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Figure 3-3. Outgoing Call Origination

Operation toward the CO is observed by removing the bridging clips and attaching a test set across tip and ring toward the open end. The test set provides loop closure. The CO detects the loop closure, attaches a dial tone generator and a digit receiver to the circuit, and establishes an audio path toward the PBX (or, in testing, toward the test set at the demarc, Figure 3-4).

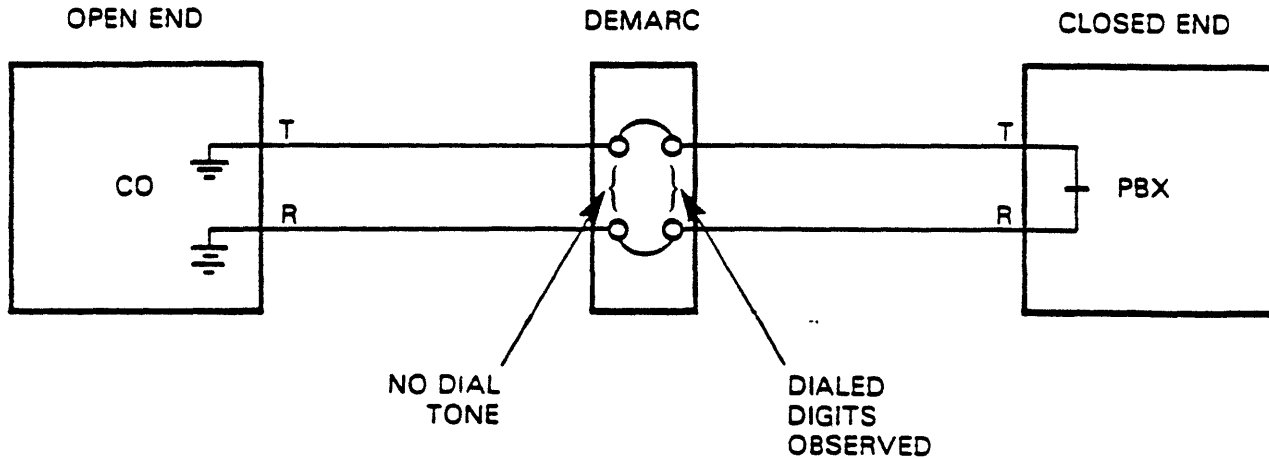


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Figure 3-4. Dial Tone from CO

Once dial tone is received by the PBX, dialing may proceed using either DTMF or dial pulse signaling as allowed by the CO (Some Central Offices are equipped to receive only dial pulse addressing. Those that are equipped to receive DTMF can also receive dial pulse.) Upon receiving the first dialed digit, the CO removes dial tone from the line (Figure 3-5).

After all digits have been dialed, the digit receiver is removed from the line and the call is routed to the distant station or switch. The audio path is extended over the outgoing facility and audible call-progress tones are returned to the calling party. Once the call is answered, business may be conducted over the audio path.

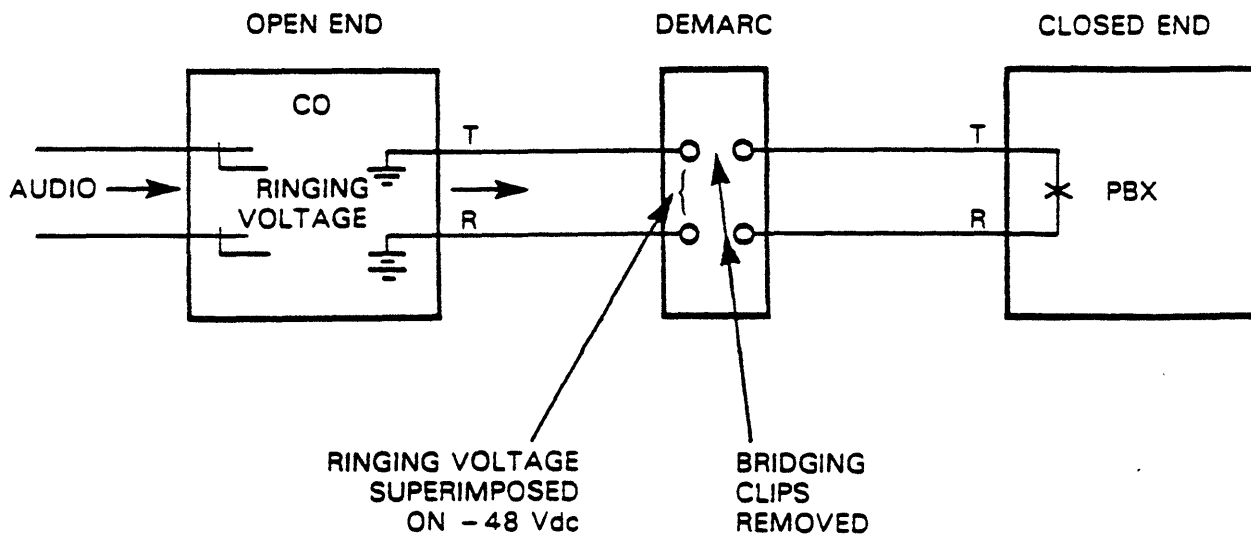


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**Figure 3-5. Dialing**

**INCOMING CALL:** An incoming call is initiated when the telephone number associated with the circuit is dialed. The CO applies ringing voltage toward the PBX (Figure 3-6). At the demarc, with the bridging clips removed, 75 to 110 Vac is observed coming from the CO on the ring lead, superimposed on -48 Vdc which is also observed at the demarc.

Ring periods of approximately 2 seconds are followed by rest periods of approximately 4 seconds.

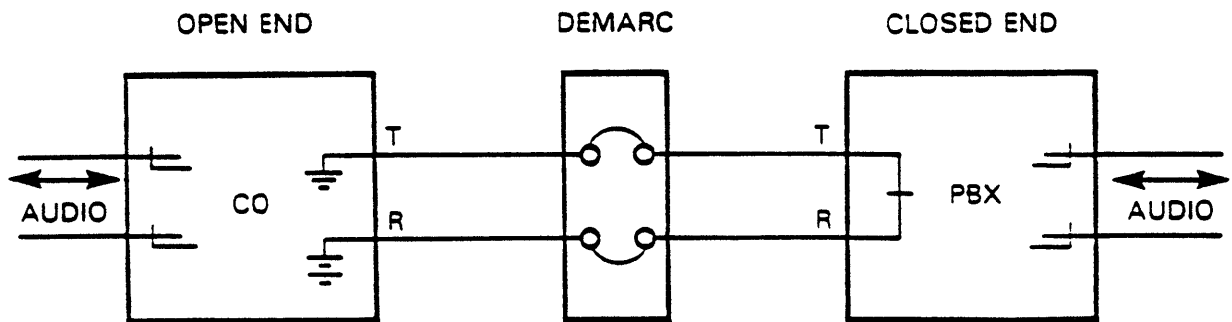


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**Figure 3-6. Incoming Ring**

## Part I Trunking

The PBX senses the incoming ring and answers the call by providing a loop closure between tip and ring. Loop current informs the CO that the call has been answered, and ringing voltage is removed (*ring tripping*). The CO then extends the audio path to the PBX (Figure 3-7).



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Figure 3-7. Answered Call

### LOOP START TO E&M INTERFACE

Loop start to E&M interface is used to extend loop start signaling over an analog carrier facility signaling system, such as Single Frequency (SF) signaling.

In the foregoing discussion of loop start signaling, signals from the CO to the PBX (and vice-versa) are carried over the wire pair between the two switching machines.

The E&M interface functions so that signaling conditions at the CO (open end) are exactly *duplicated* at the PBX (closed end).

In other words, when the CO looks toward the PBX, it will not be able to tell whether it is directly connected to the PBX by a pair of wires or by some other transmission medium. The same applies to the PBX when it looks toward the CO.

### DETAILED OPERATION OF LOOP START TO E&M INTERFACE

Detailed operation of the loop start to E&M interface will be discussed in terms of Figures 3-8a through 3-8g (outgoing call) and Figures 3-9a and 3-9b (incoming call).

The events and their associated figures are:

EVENT	FIGURE
Outgoing Call	
Idle Condition	3-8a
PBX Seizure	3-8b
CO Response	3-8c
Dial Pulse Break	3-8d
DTMF Addressing	3-8e
Occupied Circuit	3-8f
Disconnect	3-8g
Incoming Call	
Incoming Ring	3-9a
PBX Answer and Ring Tripping	3-9b

The figures contain block diagrams of an end-to-end connection that uses E&M signaling to interface loop start signaling at the open and closed ends with SF signaling over the analog carrier facility

In the diagrams, MFT stands for Metallic Facility Terminal. The MFT combines transmission and signaling functions into a single physical unit. Within the MFT, the 4-WTS provides an interface between 2-wire transmission on the drop side and 4-wire transmission on the line side. The loop to E&M box converts loop signaling to E&M signaling.

For all signaling conditions, the electrical state of the tip and ring leads in the MFT at the closed end (PBX) will be the same as the state of the tip and ring leads in the CO. Similarly, the electrical state of the tip and ring leads in the MFT at the open end (CO) will be the same as the state of the tip and ring leads in the PBX.

The symbols used for the tip and ring lead states are:

B	Battery
C	Ground
O	Open between T&R
S	Short between T&R
20 Hz	20 Hz Ringing Voltage

AFT stands for Analog Facility Terminal. The AFT interfaces 4-wire transmission on the drop side with analog carrier transmission on the line side. The E&M to SF box converts E&M signaling to SF signaling.

The status of the audio path in both directions is shown in the arrows at the top of the diagrams. The symbols used are:

I	Idle, no audio path
DT	Dial Tone
DTMF	Dual Tone Multifrequency
A	Audio or talking path

The signaling states in both directions are shown in the arrows at the bottom of the diagrams. The symbols used are:

B	Battery
C	Ground
O	Open
SF ON	Single Frequency (2600 Hz) on
SF OFF	Single Frequency (2600 Hz) off

There are two types of E&M to loop start interfaces: normal and inverted. In the figures, inverted operation is indicated in parentheses. The inverted operation is the most common form of Loop to E&M signaling, as it provides additional protection against false rings and removes SF during the talking condition.

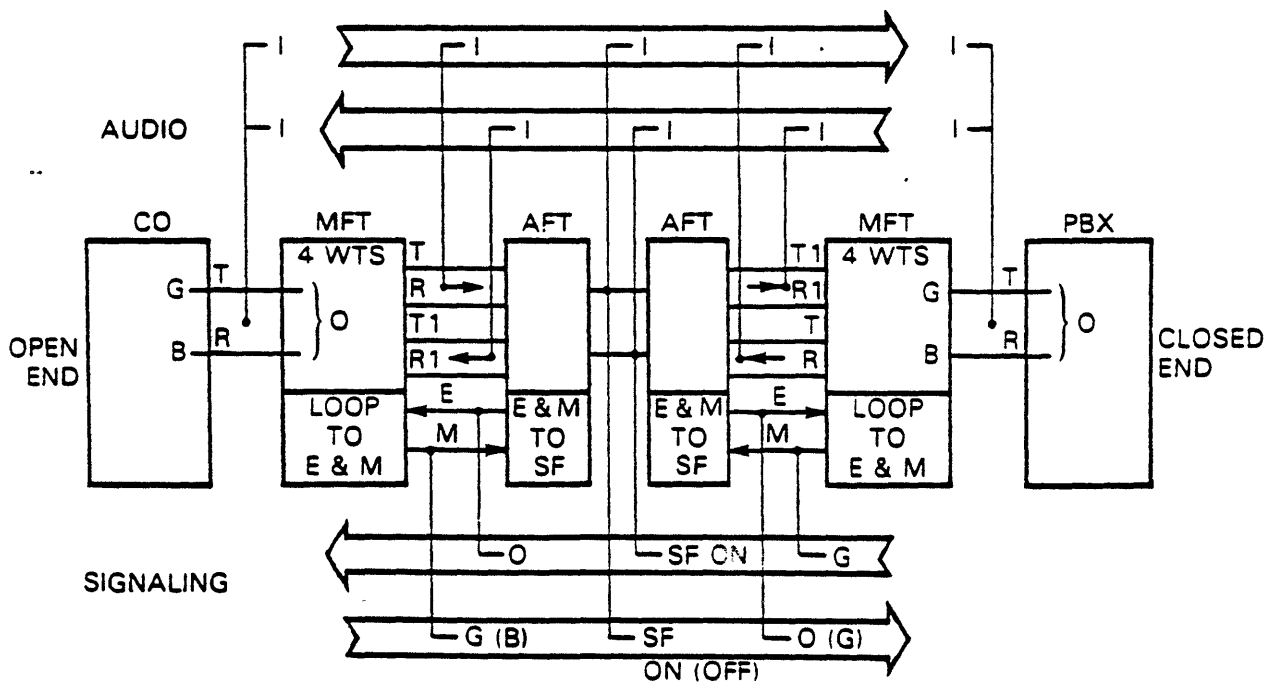
A change of state is represented by a slash between two states. For example, O/C indicates a change from open to ground.

**Part I  
Trunking**

**OUTGOING CALL (PBX ORIGATION):** An outgoing call, originated by the PBX, is discussed with reference to Figures 3-8a through 3-8g.

**Idle Condition:** In the idle condition, the tip and ring are open looking toward the PBX from the MFT. Proceeding from right to left in Figure 3-8a, the idle condition is signaled toward the facility as ground on the M lead. SF is transmitted over the analog carrier facility toward the open end AFT, where the idle condition is translated to open on the E lead toward the MFT. In the MFT, open on the E lead is translated to open on the tip and ring toward the CO, duplicating the open condition at the PBX end. The audio path from the PBX to the CO is idle.

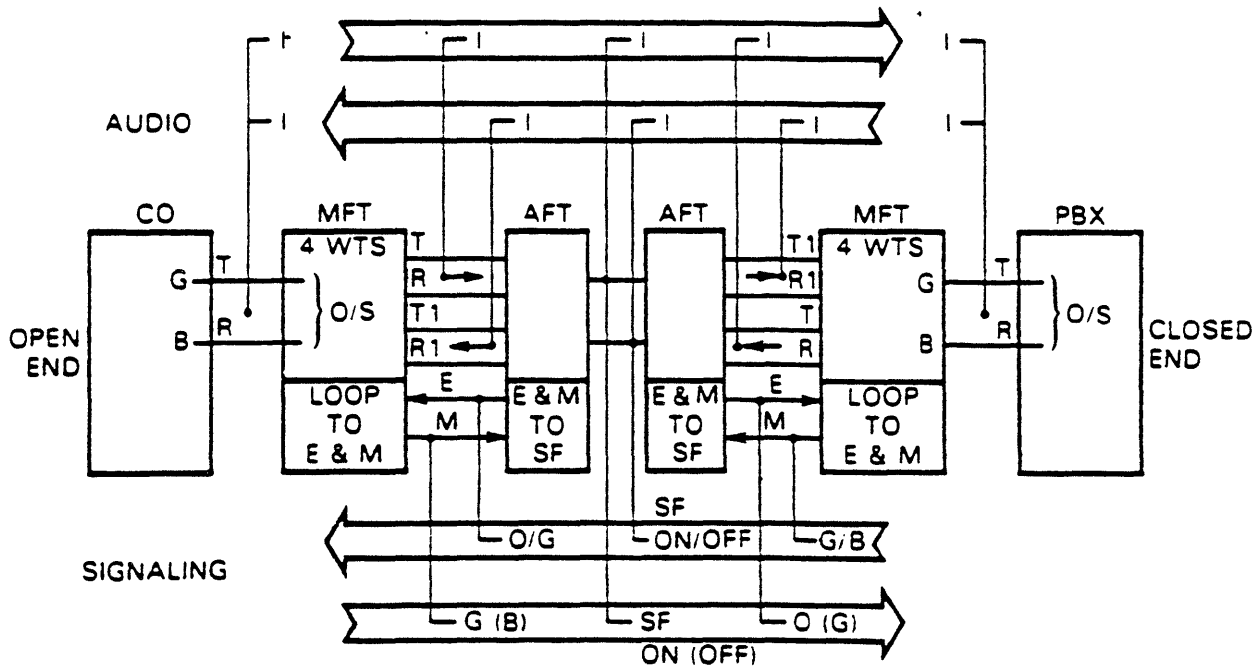
Looking toward the CO, tip is connected to ground, and ring is connected to battery. From the open end toward the closed end, the idle condition is signaled over the M lead as ground in normal operation, and as battery in inverted operation. In normal operation, this is translated to SF On over the analog carrier facility. In inverted operation, SF is Off. Signaling toward the PBX via the E lead is open for normal operation. In the loop to E&M conversion unit, E lead signaling is translated to ground on the tip lead and battery on the ring lead toward the PBX, duplicating the signaling state at the CO end. The audio path from the CO to the PBX is idle.



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Figure 3-8a. Idle Condition

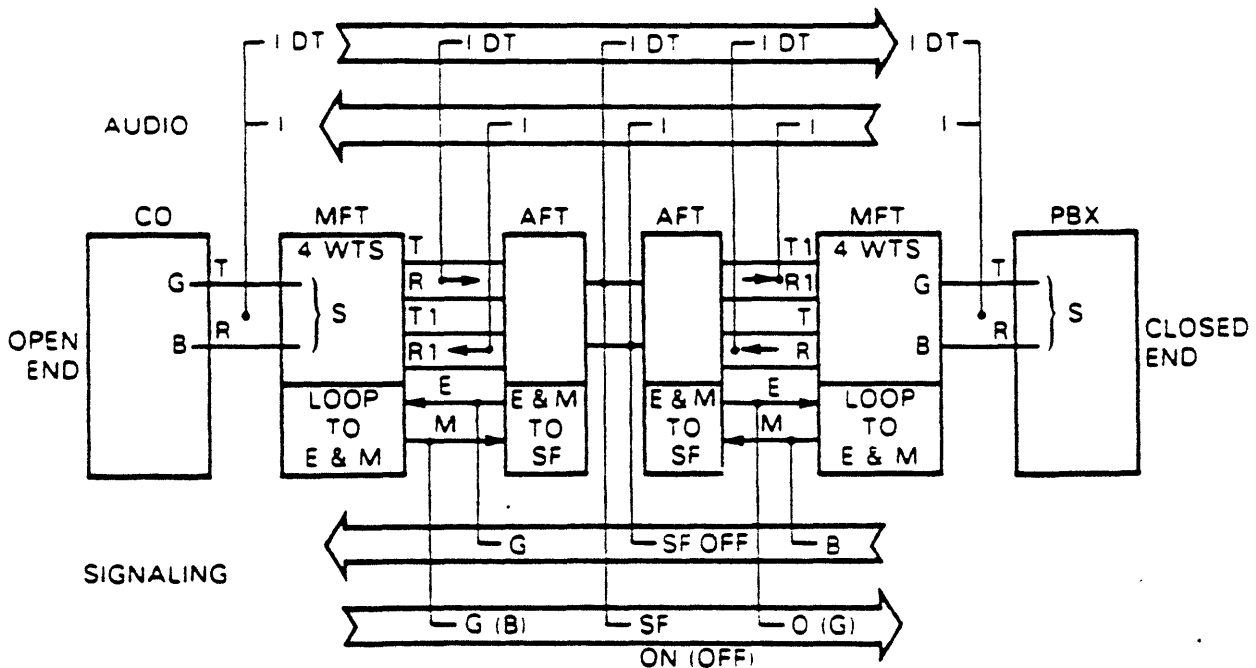
**PBX Seizure:** The PBX seizes the line by closing the loop between tip and ring. The off-hook condition is signaled toward the open end as shown in Figure 3-8b



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**Figure 3-8b. PBX Seizure**

**CO Response:** The CO responds to ground on the E lead by establishing an audio path and transmitting dial tone toward the PBX (Figure 3-8c). Note that there is no signaling occurring over the M lead — the M lead from the CO is used only to signal incoming call requests (ringing). SF over the facility remains On for normal operation and Off for inverted operation

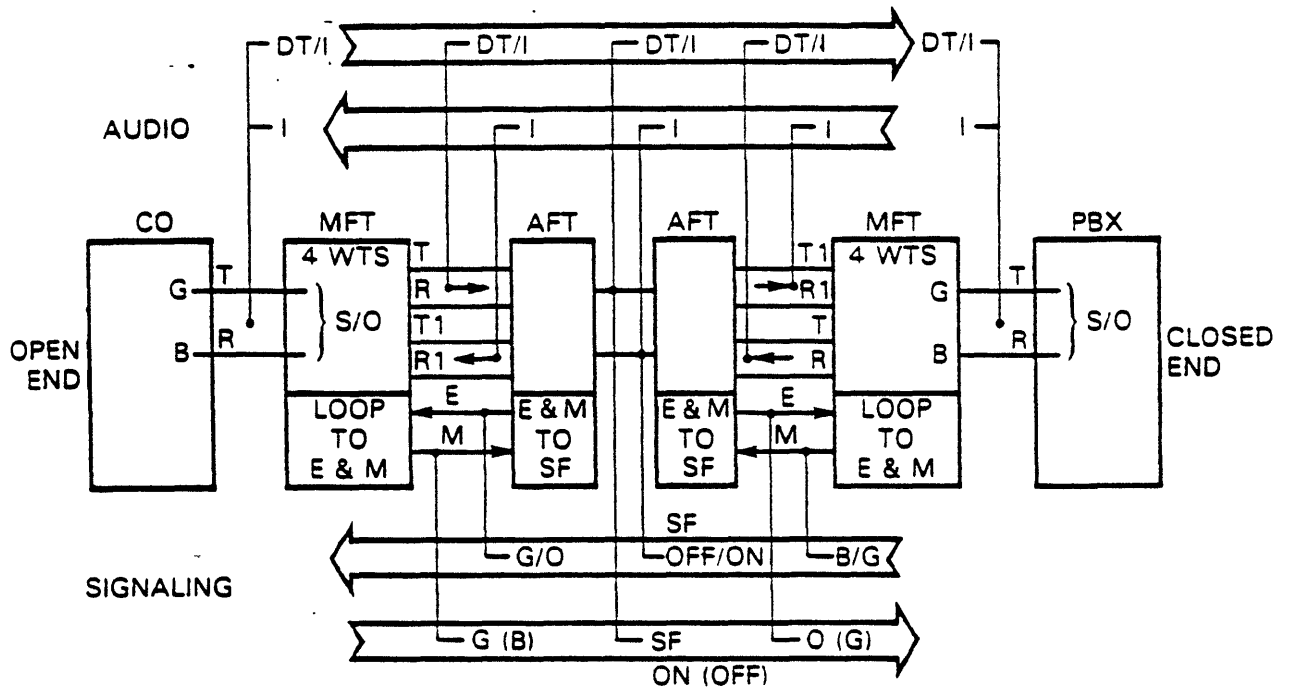


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**Figure 3-8c. CO Response**

**Part I  
Trunking**

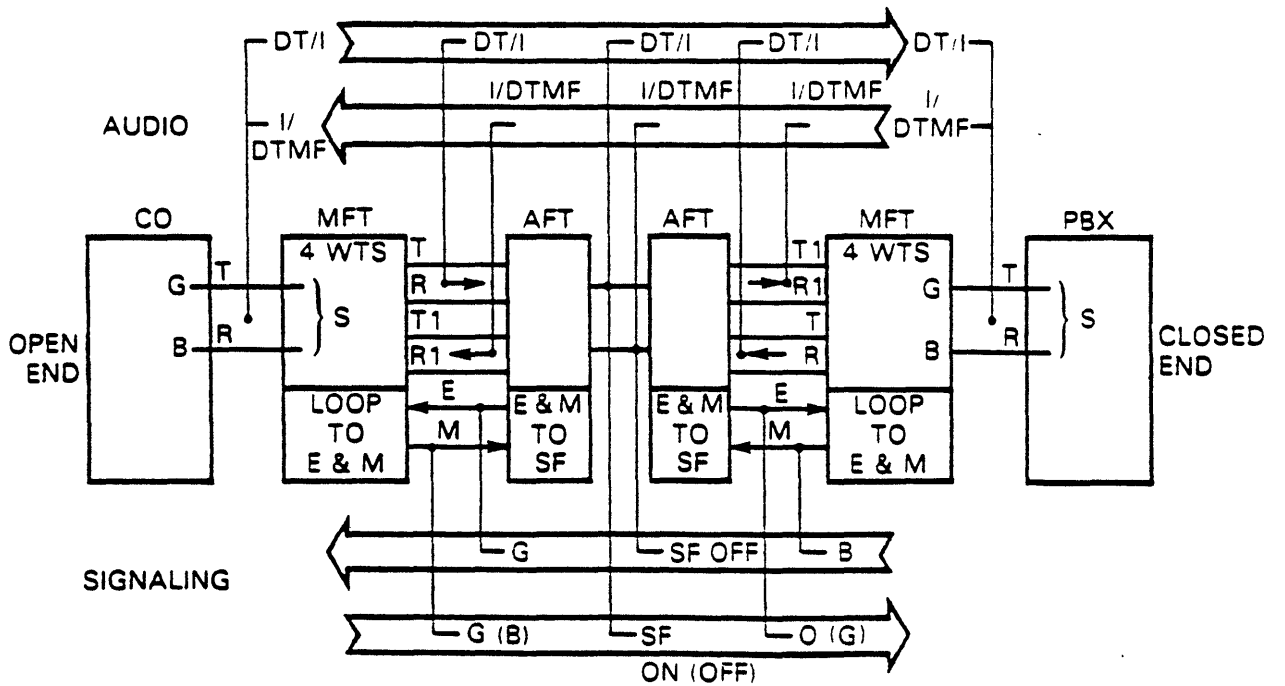
**Dial Pulse Break:** Upon receiving dial tone, the PBX sends dial pulse address information to the CO as shown in Figure 3-8d.



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Figure 3-8d. Dial Pulse Break

**DTMF Addressing:** DTMF addressing is accomplished over the audio path as shown in Figure 3-8e

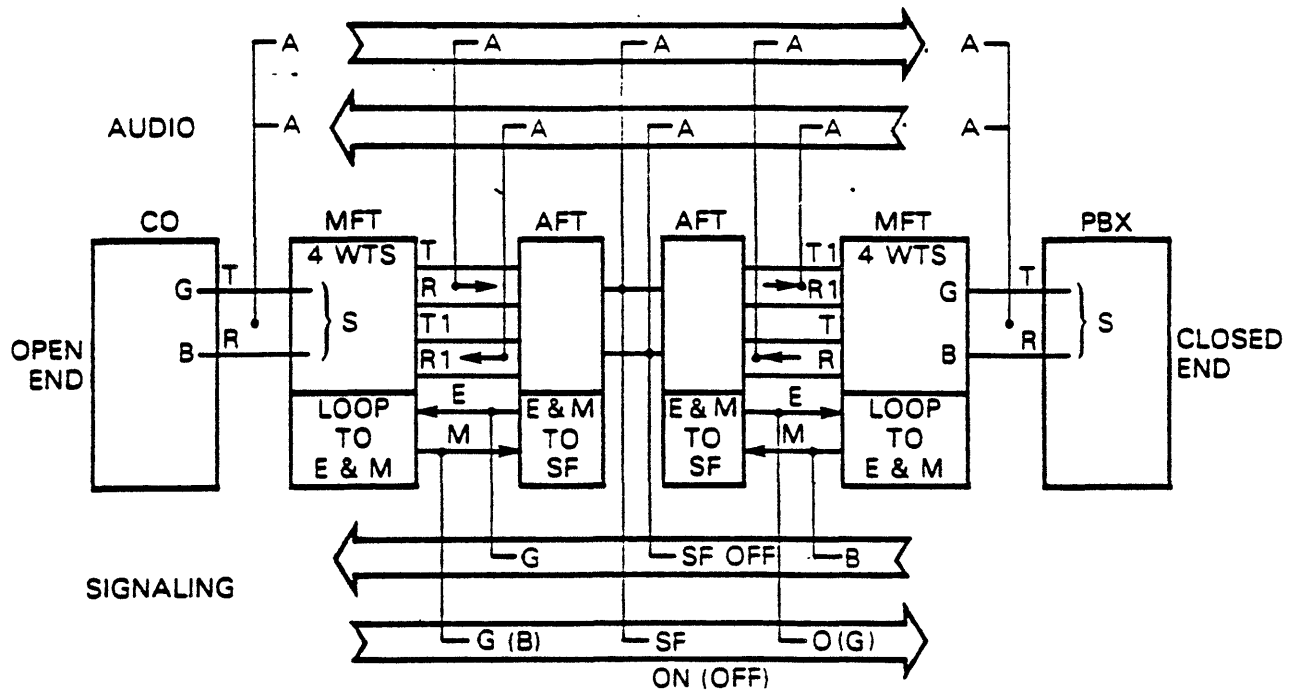


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Figure 3-8e. DTMF Addressing



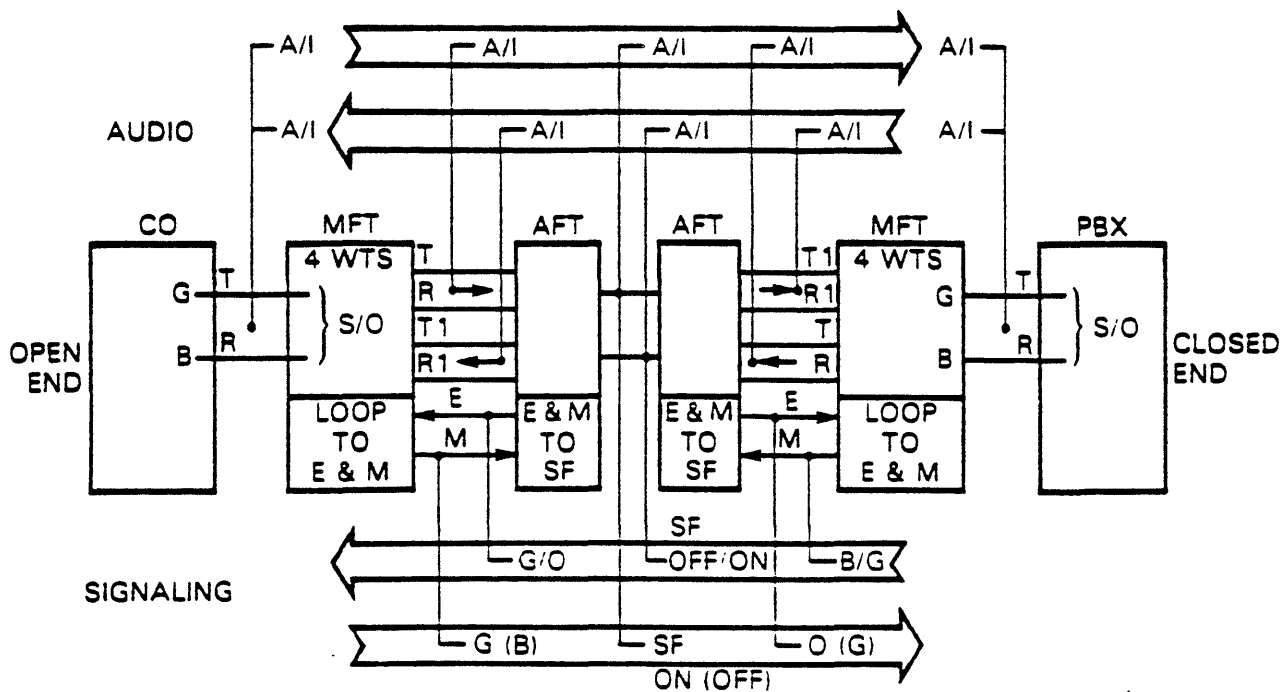
**Occupied Circuit:** The signaling states in the occupied condition are represented in Figure 3-8f



10813A034

**Figure 3-8f. Occupied Circuit**

**Disconnect:** Near-end (PBX) disconnect signaling is shown in Figure 3-8g.



10813A035

**Figure 3-8g. Disconnect**

## Part I Trunking

**INCOMING CALL (CO ORIGINATION):** An incoming call, originated by the CO, is discussed with reference to Figures 3-9a and 3-9b.

**Incoming Ring:** A call is originated by the CO by applying 20 Hz ringing voltage (75 to 110 Vac) to the ring lead, superimposed on -48 Vdc. The changes in signaling states are shown in Figure 3-9a.

In normal operation, SF is Off during the ringing period, which translates into ground on the E lead toward the closed end MFT. The MFT contains a continuous ringing supply and applies 20 Hz ringing voltage to the ring lead toward the PBX in response to the E lead ground.

Normal M lead operation can result in false ringing when SF is used for analog carrier facility signaling. If the carrier path is interrupted due to a microwave path fade or similar problem, the removal of SF will be interpreted by the E&M to SF unit as an incoming ring. A 20 Hz continuous ring will be applied toward the closed end, even though there is not an actual incoming call.

For this reason, inverted operation should be used in conjunction with SF signaling. An interruption of the carrier path when inverted operation is used will not result in ringing.

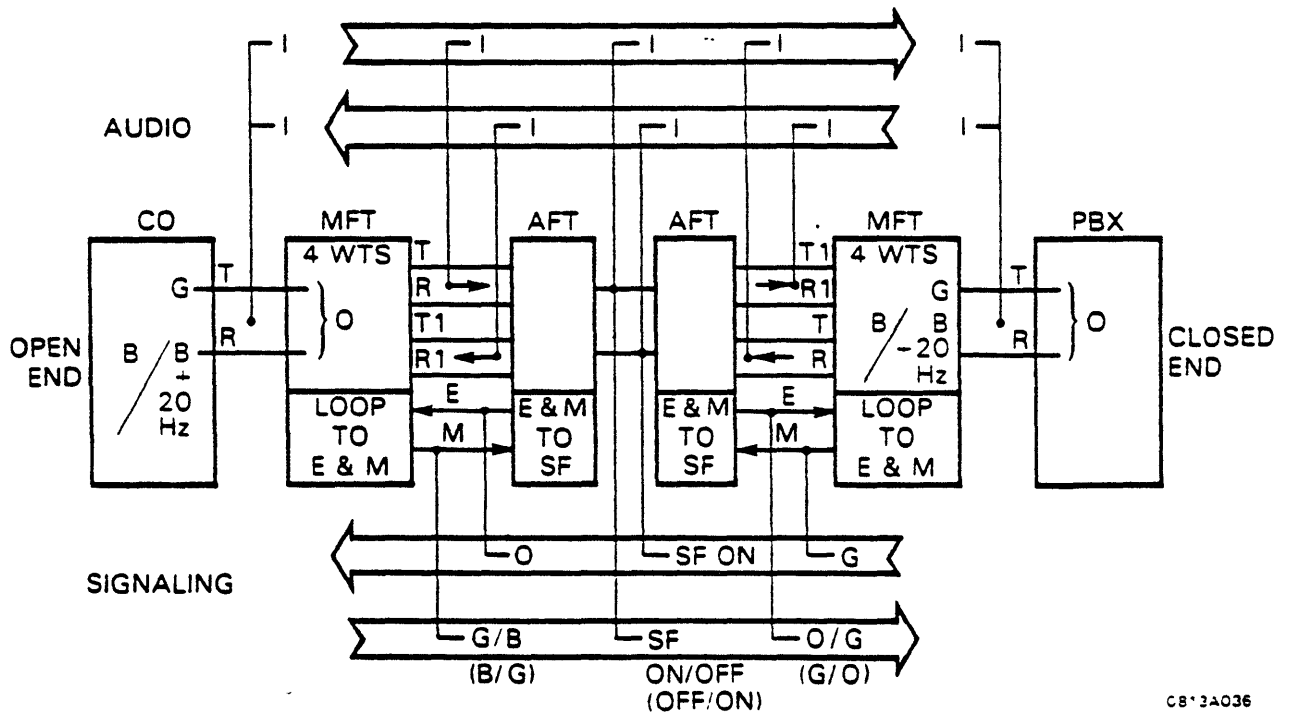
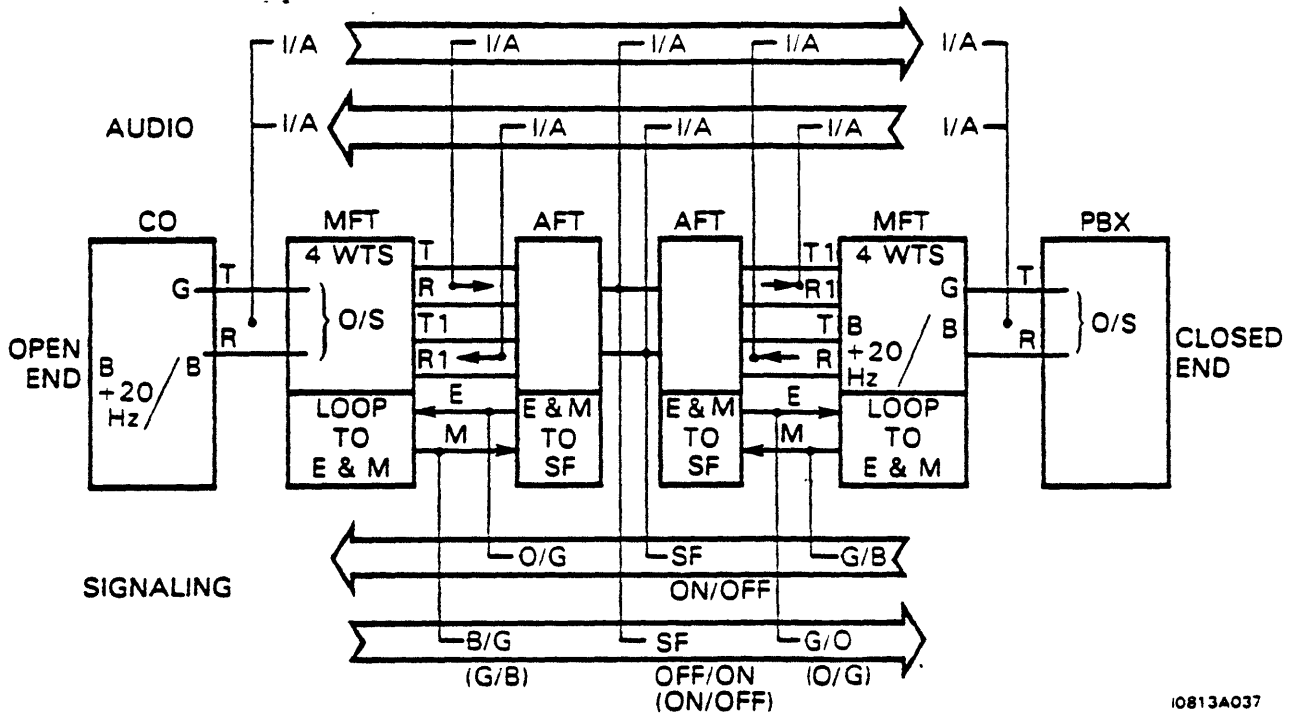


Figure 3-9a. Incoming Ring

**PBX Answer and Ring Tripping:** The PBX responds to incoming ring by providing a loop closure, which is signaled toward the CO as shown in Figure 3-9b. Upon receiving ground on the E lead, the CO trips the ringing and the call is complete.



**Figure 3-9b. PBX Answer and Ring Tripping**

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## **CHAPTER 3 REVIEW QUESTIONS**

1. What are the problems encountered in using loop start signaling over CO trunks?
2. What are the electrical states of the tip and ring leads in the idle condition in loop start operation:
  - At the CO?
  - At the PBX?
3. How does the PBX originate an outgoing call?
4. How is an incoming call originated?
5. What is false ringing and how can it be avoided?

## CHAPTER 4 GROUND START SIGNALING.

### INTRODUCTION

Ground start signaling, like loop start, uses a loop signaling format. It is used between a PBX and Central Office and has three major advantages over loop start for this application.

- The incidence of glare, simultaneous seizure of opposite ends of a circuit, is reduced
- The CO sends positive signaling (in addition to dial tone) when digit collection equipment has been attached.
- The CO can signal the PBX when the distant end disconnects.

This chapter provides an overview of ground start operation. The overview is followed by a detailed discussion of ground start signaling with reference to block diagrams.

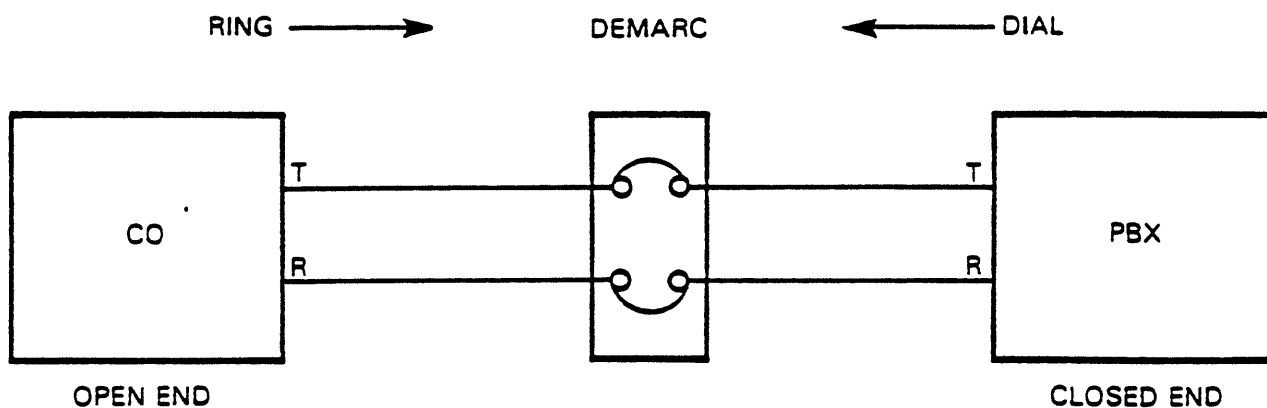
The E&M to ground start interface is presented in terms of the signaling conversions that take place.

Appendix B contains a discussion of the overall operation of ground start circuitry and ground start to E&M interface circuitry. The discussion is conducted with reference to simplified circuit schematics.

### GROUND START OVERVIEW

This section provides an overview of ground start signaling over a 2-wire circuit between a PBX and a Central Office. Figure 4-1 illustrates the connection. As indicated in the figure, an incoming call to the PBX is initiated by ringing from the CO. The PBX dials outgoing calls. This is called one-way ring, one-way dial operation.

In this section, ground start is discussed with reference to the idle condition, an outgoing (PBX originated) call, an incoming (CO originated) call, and a call disconnect.



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Figure 4-1. PBX/Central Office Connection

## Part I Trunking

**IDLE CONDITION:** During the idle state, the ring lead is attached to the battery and the tip lead is open in the Central Office. In the PBX, the ring lead is open and the tip lead is attached to the battery. As will become apparent, this configuration provides for more signaling states than are available in loop start. Table 4-1 compares the idle condition for loop start and ground start.

Table 4-1. Loop Start vs. Ground Start - Idle Condition

SIGNALING	CO (OPEN END)	PBX (CLOSED END)
Loop Start		
Tip	GND	Open
Ring	- 48 Vdc	Open
Ground Start		
Tip	Open	- 48 Vdc
Ring	- 48 Vdc	Open

**OUTGOING CALL:** An outgoing call is initiated when the PBX applies a ground to the ring, causing current to flow over the ring lead between the PBX and the CO. The CO recognizes the grounded ring lead as a service request and attaches a digit receiver in preparation for receiving address information. The CO informs the PBX that it is ready to receive address information by placing a ground on the tip. The CO also provides dial tone.

Upon receipt of tip ground from the CO, the PBX simultaneously removes its ring ground toward the CO and provides a loop closure between the tip and ring leads.

The tip ground provides the PBX with a positive indication that the CO is prepared to receive digits. You will recall that in loop start operation, a PBX not equipped to recognize dial tone waits for a specified period of time prior to sending digits, on the assumption that the CO will have attached a digit receiver by the end of the wait period. Ground start, therefore, provides a more positive start-dialing signal than does loop start.

If the PBX is non-senderized, the subscriber hears dial tone from the CO and may proceed to send address information.

Depending on the type of digit recognition provided by the CO, the PBX (or the subscriber) will then proceed to output the address information as either dial pulse or DTMF signals. Upon receipt of the first digit, the CO removes dial tone from the line. Once all digits have been outputted and connected by the CO, an audio path is established between the call originator and the CO.

Any further call progress signals are transmitted directly to the call originator as audible signals.

**INCOMING CALL:** An incoming call from the CO is initiated when the CO grounds the tip lead toward the PBX. The PBX sees the tip ground as a busy trunk and restricts it from being selected for an outgoing call.

In loop start operation, CO seizure consists of the application of ringing voltage toward the PBX. The PBX can seize the circuit during the rest period between rings, resulting in a glare condition. In ground start, tip ground from the CO provides more positive trunk seizure than does the application of ringing voltage, reducing the possibility of glare.

The CO proceeds to apply interrupted ringing (2 seconds ring, 4 seconds rest) on the ring lead toward the PBX. The PBX sees the incoming ring as an incoming call request and places a loop closure across the tip and ring leads.

The CO detects the loop closure, disconnects the ringing supply, and establishes an audio path over the tip and ring leads. The PBX may then route the call to a user-selected location, to an operator for further routing, or to dial tone to inform the caller that it is ready to receive further address information (non-senderized operation).

As with loop start, only DTMF may be used for incoming non-senderized operation.

If incoming addressing is to be allowed, it must be recognized that power losses over incoming connections may cause DTMF levels that are too low to be detected by the PBX. This is particularly true of tandem tie trunk connections operating in the non-senderized mode. In this case, the DTMF pulses may have traversed several links and traveled a considerable distance.

**CALL DISCONNECT:** Either party may disconnect the call by going on-hook. If the PBX disconnects first, the loop closure will be removed toward the CO. The CO will see this as an absence of current over the facility and will disconnect the call, returning an open tip toward the PBX. The PBX will see the open tip as an idle trunk and return the trunk to the idle state, making it available for outgoing selection.

If the distant end disconnects first, the CO will open the tip lead, causing loop current to cease. The PBX will sense the tip ground removal and will remove the loop closure and return to the idle condition.

## **DETAILED OPERATION OF GROUND START SIGNALING**

The signaling states of ground start operation will be discussed with reference to two viewpoints from the demarc: looking toward the open end (the CO) and toward the closed end (the PBX).

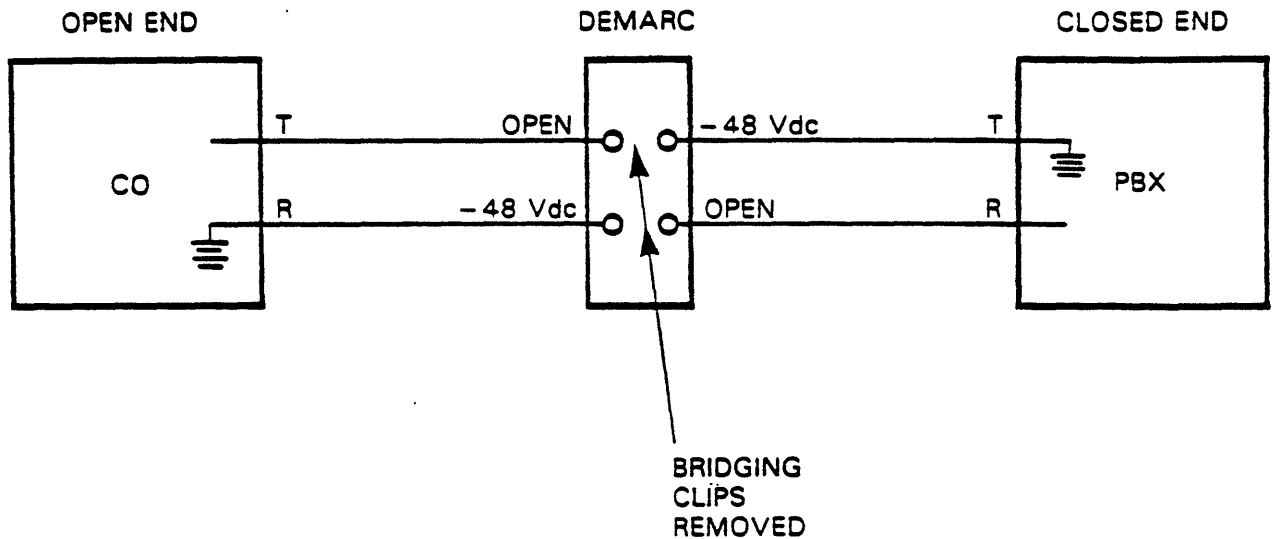
The discussion will be with reference to the following events and their associated figures.

<b>EVENT</b>	<b>FIGURE</b>
Idle Condition	4-2
PBX Seizure	4-3
PBX Loop Closure	4-4
PBX Address Signaling	4-5
CO Response to PBX Seizure	4-6
Addressing Toward CO	4-7
Incoming Seizure and Ringing	4-8
Ring Tripping	4-9

Although this section is not intended as a guide to circuit testing, the signaling operations are described in much the same manner as they would be observed during testing.

## Part I Trunking

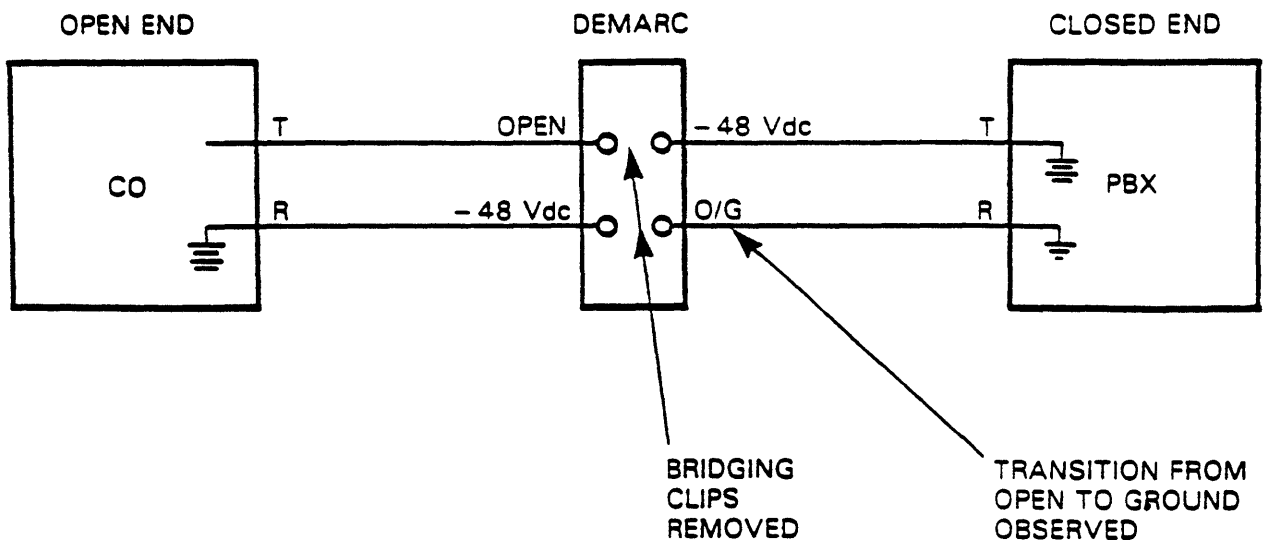
**IDLE CONDITION:** The idle condition is represented in Figure 4-2. The bridging clips have been removed to isolate the open and closed ends. Looking toward the closed end (the PBX),  $-48\text{ Vdc}$  is observed on the tip lead, and the ring lead is open. Looking toward the open end (the CO),  $-48\text{ Vdc}$  is observed on the ring lead, and the tip lead is open.



10813A039

Figure 4-2. Idle Condition

**OUTGOING CALL, CLOSED END:** An outgoing call is initiated when the PBX places ground on the ring lead. With the bridging clips removed, the transition from open to ground is observed on the ring lead looking toward the PBX. (Figure 4-3.)

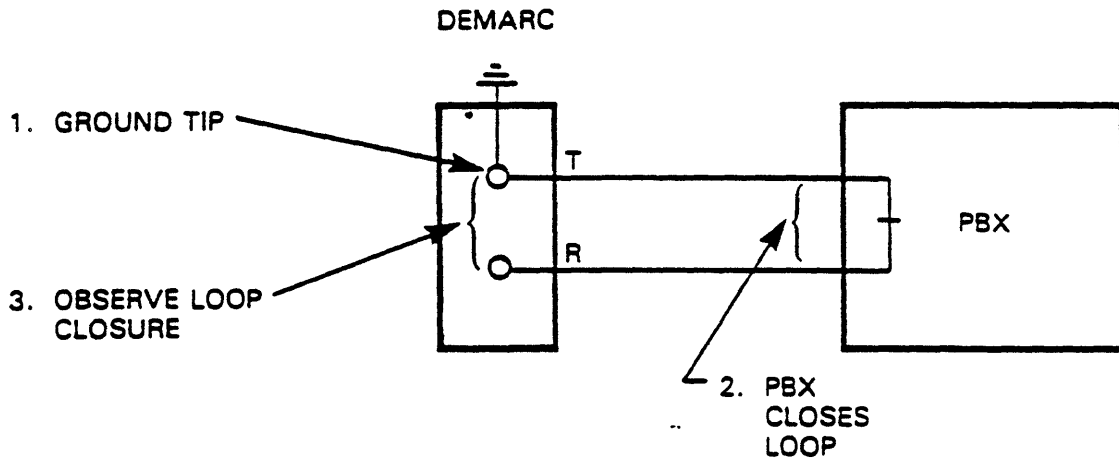


10813A040

Figure 4-3. PBX Seizure



The CO response to outgoing seizure may be simulated by placing a ground on the tip lead at the demarc toward the PBX. The PBX responds by removing the ground from the ring lead and providing a loop closure between tip and ring. The loop closure may be observed at the demarc as a conducting path between tip and ring. (Figure 4-4.)

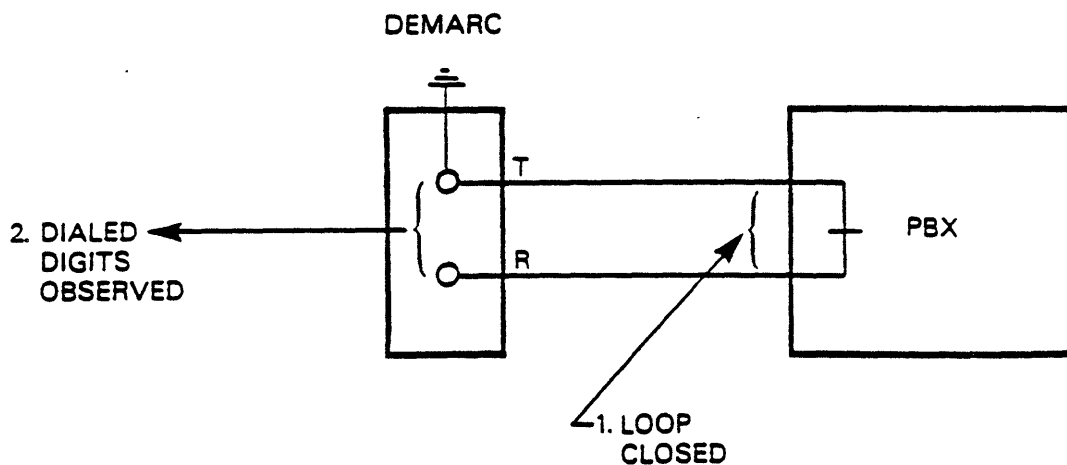


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**Figure 4-4. PBX Loop Closure**

At this point in an actual call, tip is grounded, ring remains connected to battery in the CO, and the CO is prepared to receive address signaling.

Using the proper test equipment, outgoing dial pulse or DTMF address signaling may be observed at the demarc. (Figure 4-5.)

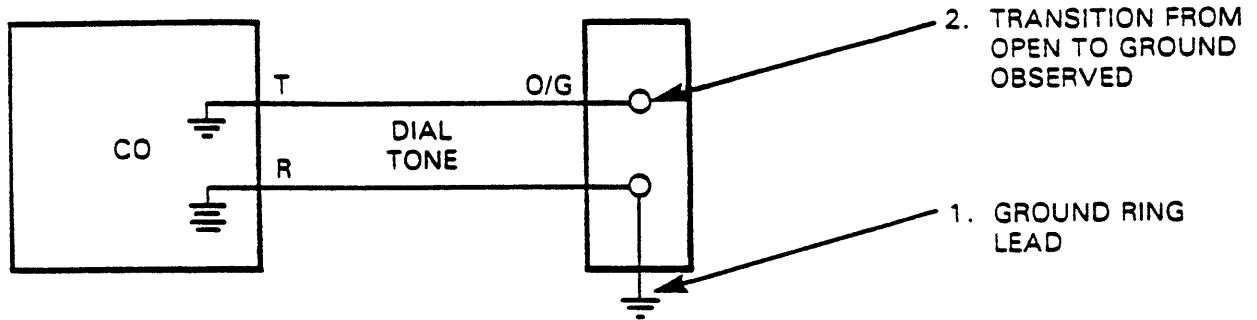


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**Figure 4-5. PBX Address Signaling**

**Part I  
Trunking**

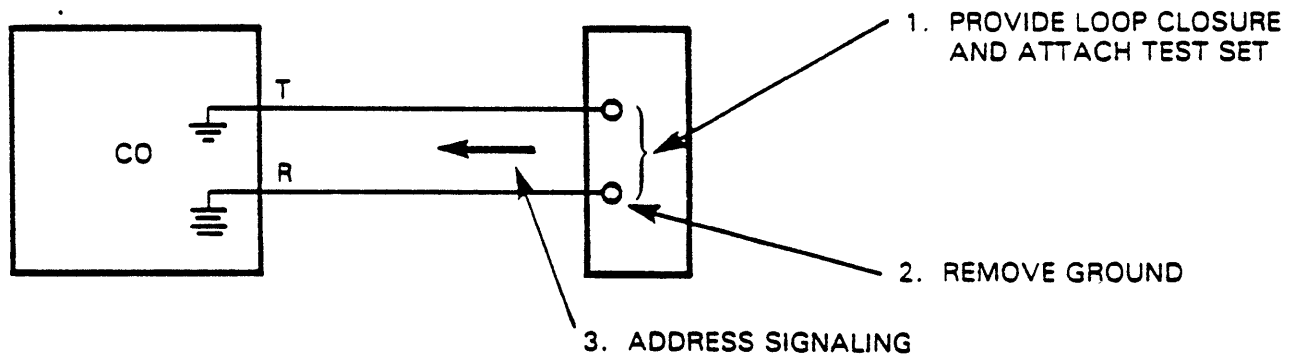
**OUTGOING CALL, OPEN END:** With the bridging clips removed, a ground is placed on the ring lead toward the CO to initiate a call request. The tip lead is observed to go from open to ground, indicating that the CO is ready to receive address information. Dial tone may also be heard on the test set. (Figure 4-6.)



10813A043

**Figure 4-6. CO Response to PBX Seizure**

In order to simulate PBX operation, a loop closure is placed across tip and ring, and the ring ground is removed. Using the correct digit format (dial pulse or DTMF), test calls may now be placed from the demarc to a test location. (Figure 4-7.)



10813A044

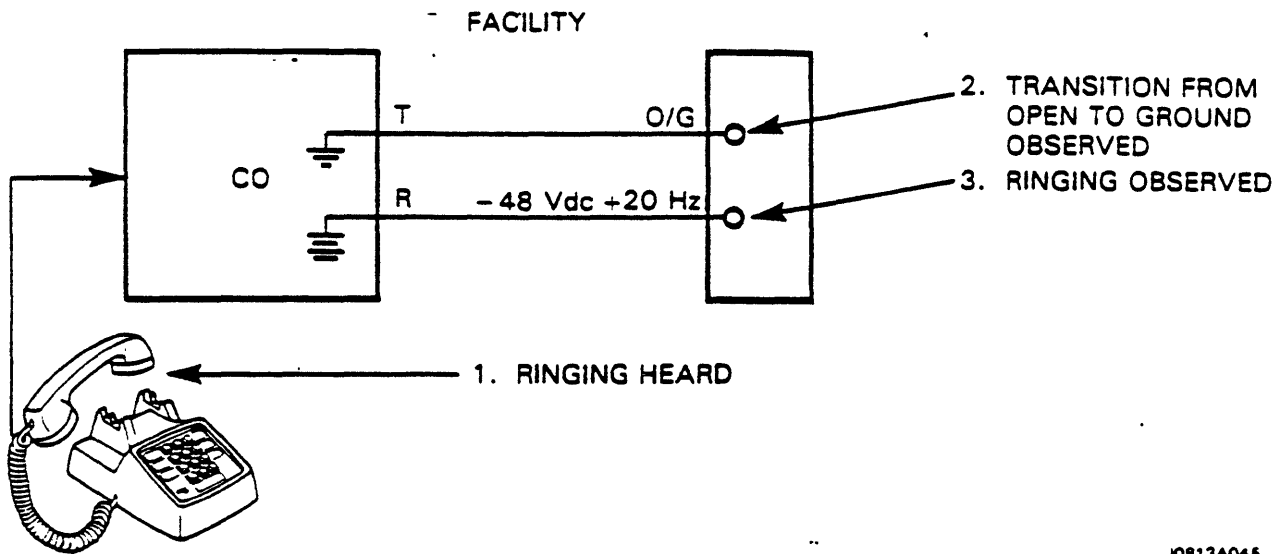
**Figure 4-7. Addressing Toward CO**

**INCOMING CALL, OPEN END:** To make an operational check on an incoming call, the facility access number is dialed from a telephone not connected to the circuit under test.

Once the call has been completed to the distant switch and ringing is heard on the originating telephone, the tip lead from the CO is observed to go from open to ground at the demarc

The ring lead is checked to ensure that interrupted ringing (2 seconds ring, 4 seconds rest) is initiated, superimposed on - 48 Vdc.

Several ringing cycles are observed to ensure that there are no conditions that would cause the ring to trip prematurely. (Figure 4-8.)

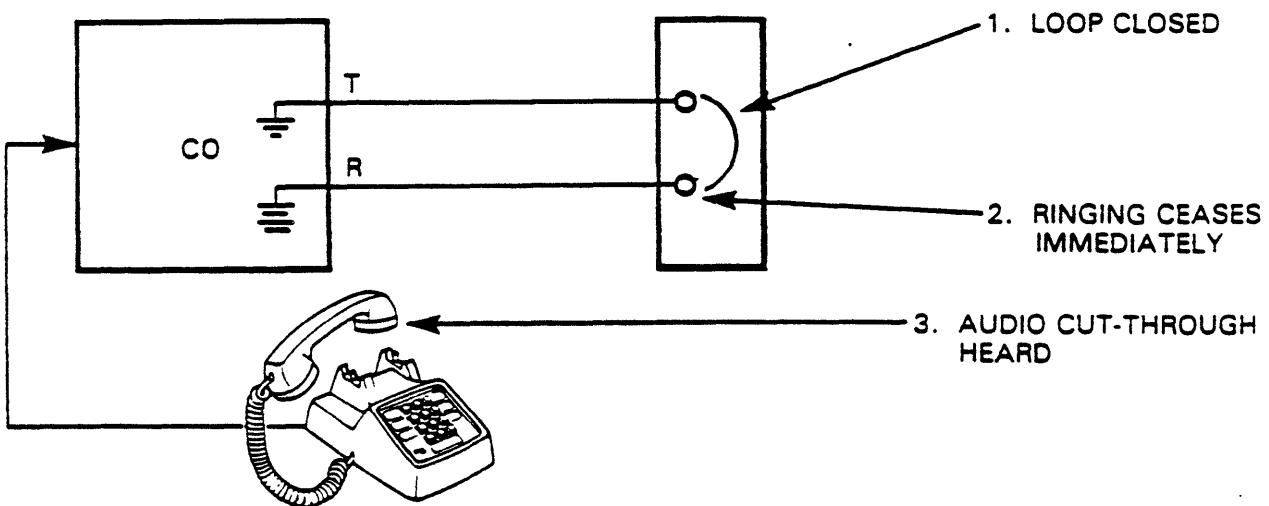


10813A045

**Figure 4-8. Incoming Seizure and Ringing**

During the ringing cycle, a loop closure is placed across the tip and ring leads in order to test ring tripping (Figure 4-9). Ringing should cease immediately, and an audio path between the facility and the originating telephone is established. This tests the pre-trip circuitry.

The call is re-established. This time, the loop closure across tip and ring is made during the rest period. Audio cut-through between the facility and the originating telephone should occur immediately, and further ringing should not occur.



10813A046

**Figure 4-9. Ring Tripping**

**CALL DISCONNECT:** Several test calls are made; after each call, tip and ring toward both ends are checked to ensure that they have returned to the idle state.

## Part I Trunking

### GROUND START TO E&M INTERFACE

The ground start to E&M signaling interface is used to extend ground start signaling over an analog carrier facility system, such as Single Frequency (SF) signaling.

In the foregoing discussion of ground start signaling, signals from the CO to the PBX (and vice-versa) are carried over the wire pair between the two switching machines.

The E&M interface functions so that signaling conditions at the CO (open end) are *exactly duplicated* at the PBX (closed end).

In other words, when the CO looks toward the PBX, it will not be able to tell whether it is directly connected to the PBX by a pair of wires or by some other transmission medium. The same applies to the PBX when it looks toward the CO.

### DETAILED OPERATION OF GROUND START TO E&M INTERFACE

Detailed operation of the ground start to E&M interface will be discussed with reference to Figures 4-10a through 4-10f (outgoing call) and Figures 4-11a through 4-11d (incoming call). Call disconnect is shown in Figures 4-12a and 4-12b.

The figures contain block diagrams of an end-to-end connection that uses E&M signaling to interface ground start signaling at the open and closed ends with SF signaling over the analog carrier facility.

In the diagrams, MFT stands for Metallic Facility Terminal. The MFT combines transmission and signaling functions into a single physical unit. Within the MFT, the 4 WTS provides an interface between 2-wire transmission on the drop side and 4-wire transmission on the line side. The loop to E&M box converts loop signaling to E&M signaling.

For all signaling conditions, the electrical state of the tip and ring leads in the MFT at the closed end (PBX) will be the same as the state of the tip and ring leads in the CO. Similarly, the electrical state of the tip and ring leads in the MFT at the open end (CO) will be the same as the state of the tip and ring leads in the PBX.

The symbols used for the tip and ring lead states are:

B	Battery
C	Ground
O	Open between T&R
S	Short between T&R
20 Hz	20 Hz Ringing Voltage

AFT stands for Analog Facility Terminal. The AFT interfaces 4-wire transmission on the drop side with analog carrier transmission on the line side. The E&M to SF box converts E&M signaling to SF signaling.

The status of the audio path in both directions is shown in the arrows at the top of the diagrams. The symbols used are:

I	Idle, no audio path
DT	Dial Tone
DTMF	Dual Tone Multifrequency
A	Audio or talking path

The signaling states in both directions are shown in the arrows at the bottom of the diagrams. The symbols used are:

B	Battery
C	Ground
O	Open
SF ON	Single Frequency (2600 Hz) on
SF OFF	Single Frequency (2600 Hz) off

A transition between states is illustrated by a slash. For example, the transition between open and ground is shown as O/G

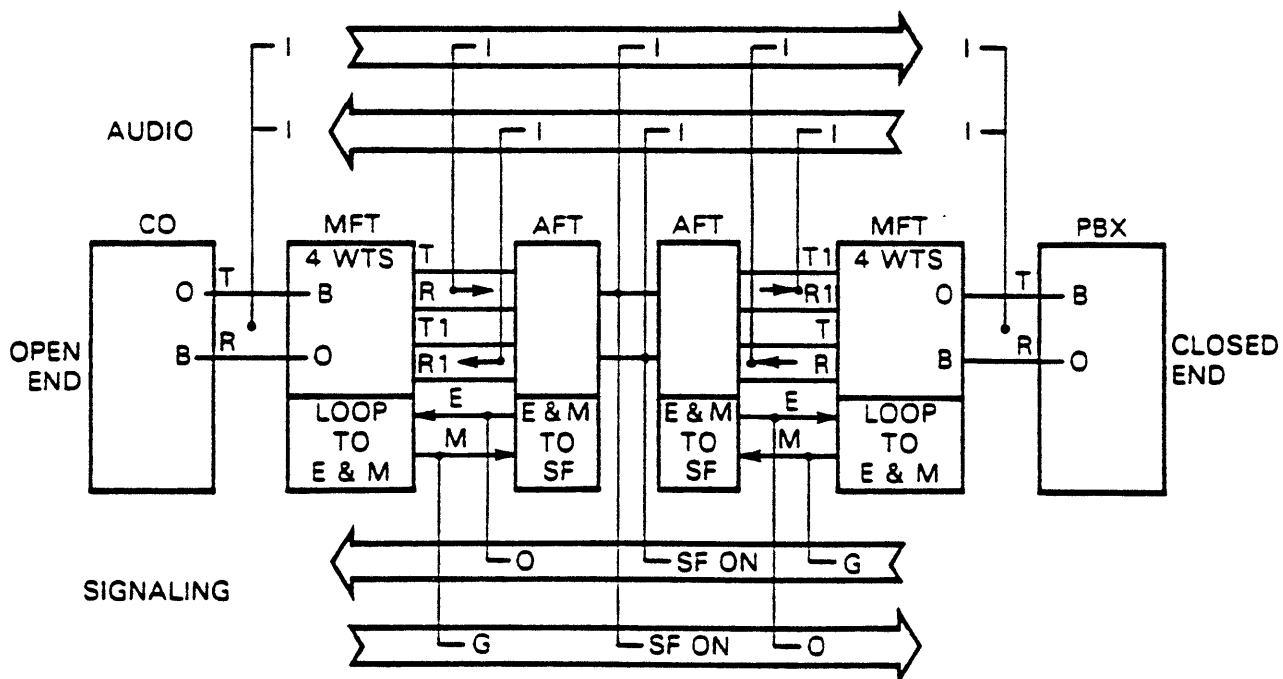
There are two types of ground start-to E&M interfaces: wink-on-ring and ring-after-ring. The differences between the two will be discussed in relation to an incoming (CO originated) call.

**OUTGOING CALL (PBX ORIGATION):** An outgoing call is discussed with reference to Figures 4-10a through 4-10f. The call is broken into a series of events, represented by the following figures:

EVENT	FIGURE
Idle Condition	4-10a
PBX Seizure	4-10b
CO Response	4-10c
DTMF Addressing	4-10d
Dial Pulse Addressing	4-10e
Occupied Circuit	4-10f

**Idle Condition:** In the idle condition, the tip is connected to battery and the ring is open, looking toward the PBX from the MFT. Proceeding from right to left in Figure 4-10a, the M lead is grounded, SF is On, and the E lead is open. The open E lead translates to battery on the tip and open on the ring toward the CO, duplicating the signaling conditions that the CO would see if it were looking directly at the PBX.

At the CO end, the tip lead is open and the ring lead is connected to battery in the idle condition. Going from left to right toward the PBX, the M lead is grounded, SF is On, and the E lead toward the closed-end MFT is open. The open E lead translates to open on the tip and battery on the ring toward the PBX, duplicating the signaling conditions that the PBX would see if it were looking directly at the CO.

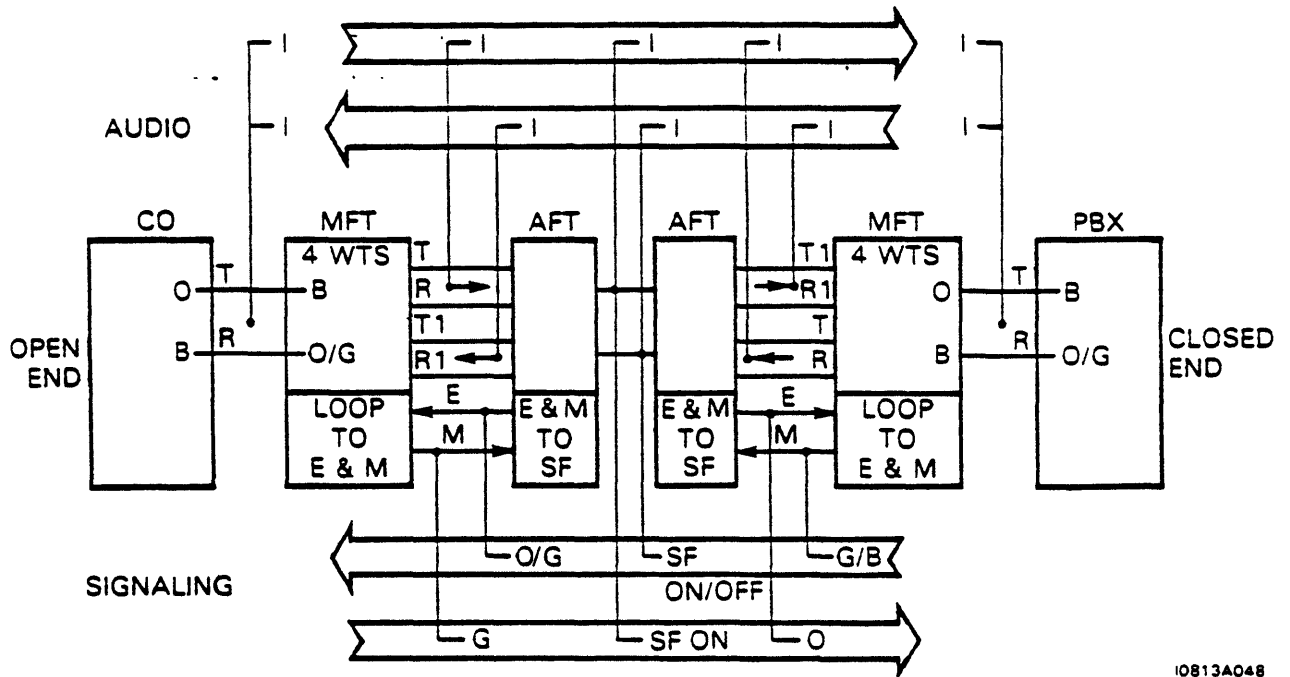


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Figure 4-10a. Idle Condition

## Part I Trunking

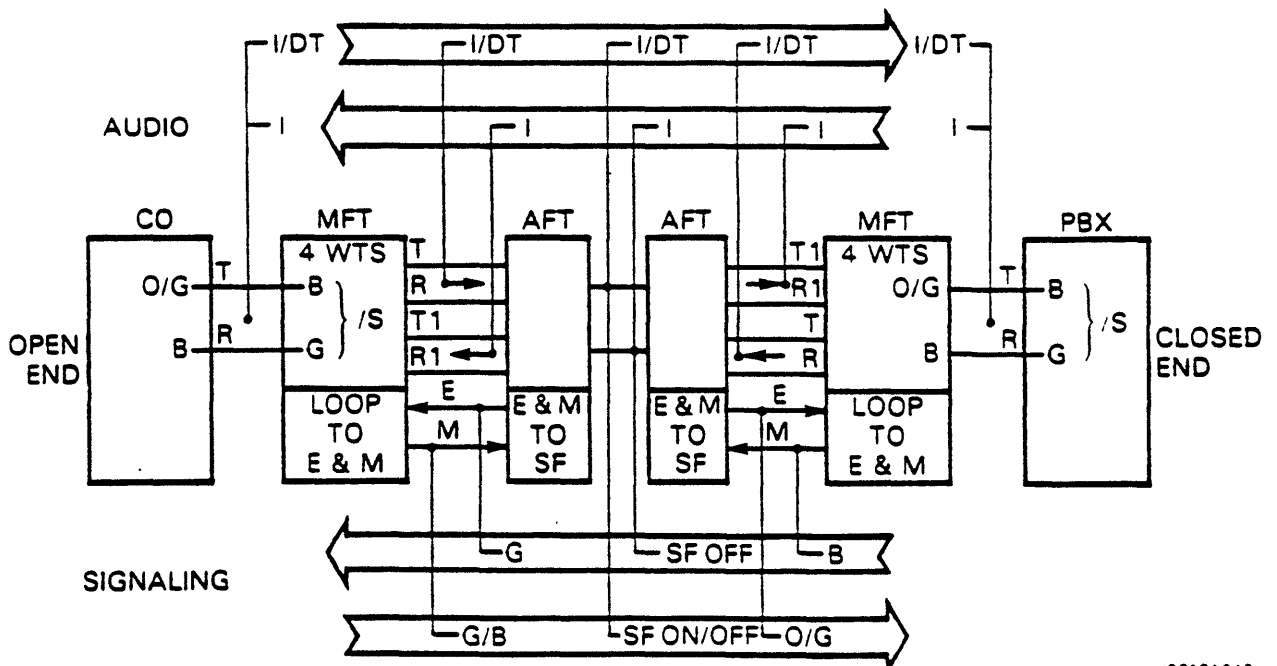
**PBX Seizure:** The PBX seizes the circuit by placing a ground on the ring lead. The corresponding signaling toward the CO is shown in Figure 4-10b.



10813A048

Figure 4-10b. PBX Seizure

**CO Response:** The CO recognizes the seizure via the ground on the ring lead and responds by grounding the tip lead toward the MFT (Figure 4-10c.) In the MFT, the loop between tip and ring is closed simultaneously with the removal of ring ground. The M lead toward the AFT goes from ground to battery, SF is turned Off, and the E lead toward the MFT goes from open to ground. In the closed-end MFT, the tip lead goes from open to ground. Dial tone is transmitted toward the PBX over the audio path.

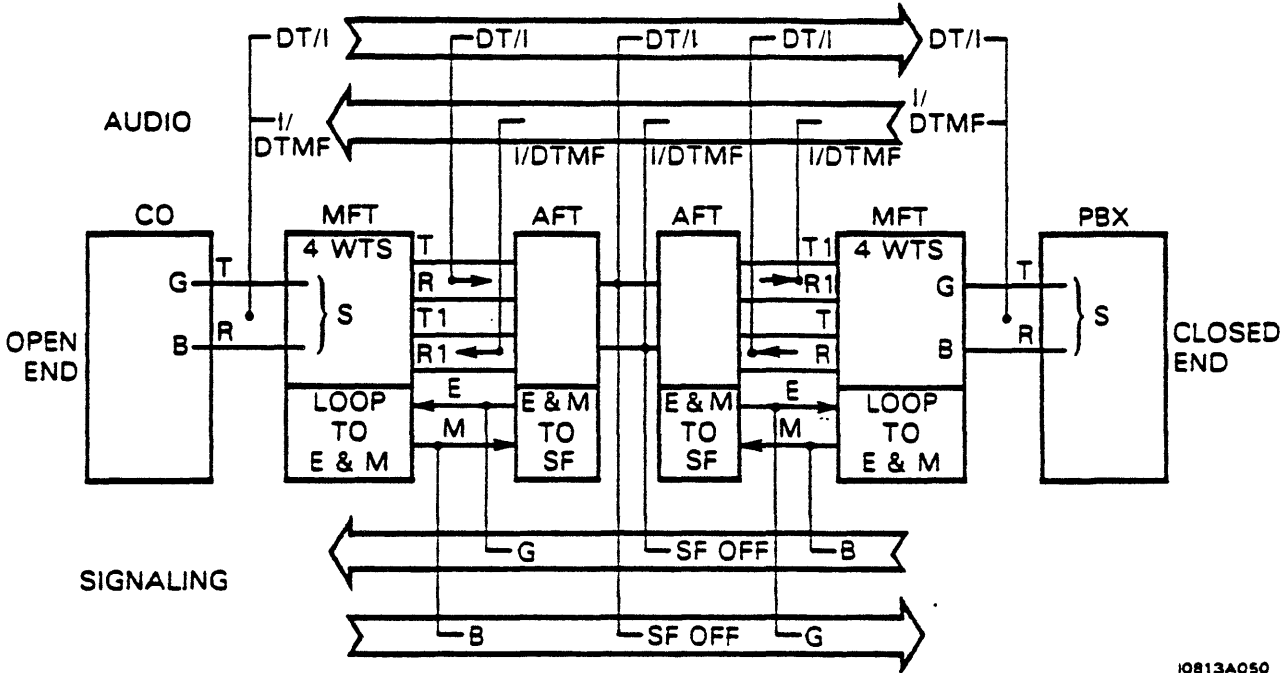


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Figure 4-10c. CO Response

In the PBX, the loop between tip and ring is closed simultaneously with the removal of ring ground.

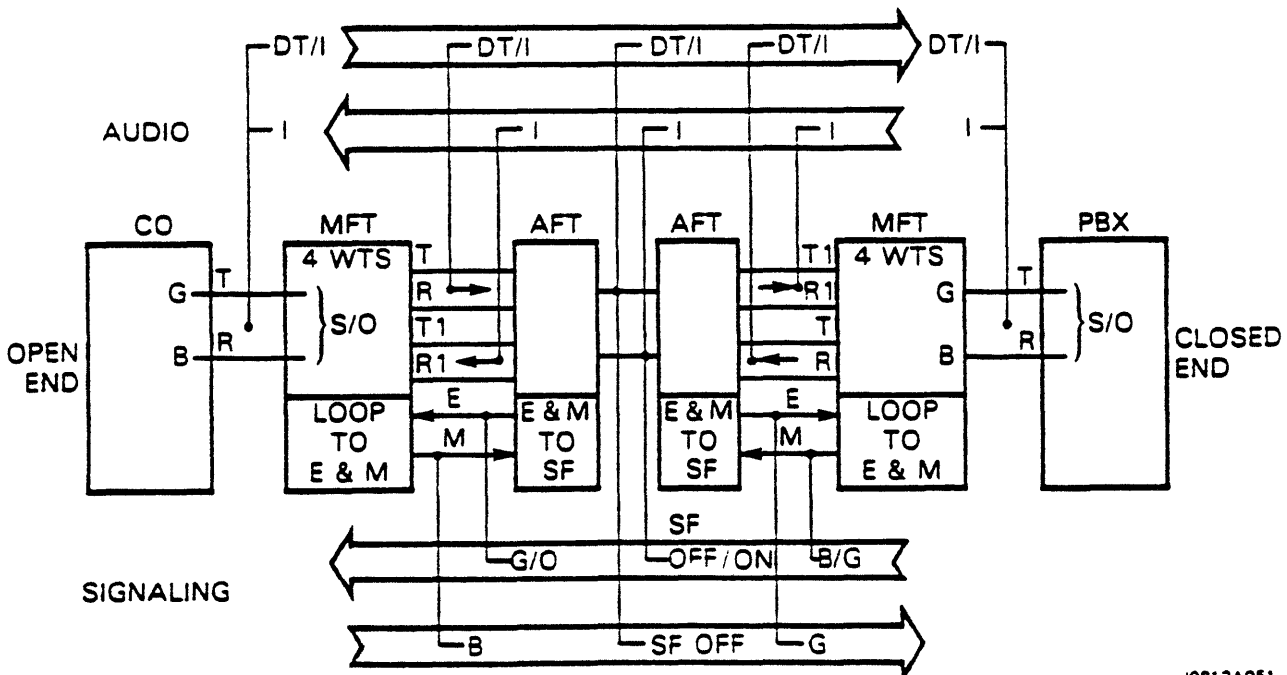
**DTMF Addressing:** The PBX recognizes the grounded tip as the start dialing signal in senderized operation. In non-senderized operation, the calling party recognizes dial tone as the start dialing signal. DTMF address digits are transmitted to the CO over the audio path, and the CO removes dial tone upon receipt of the first digit. (Figure 4-10d.)



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**Figure 4-10d. DTMF Addressing**

**Dial Pulse Addressing:** As in loop start signaling, dial pulse addressing is transmitted toward the CO by opening and closing the loop in the PBX. (Figure 4-10e.)

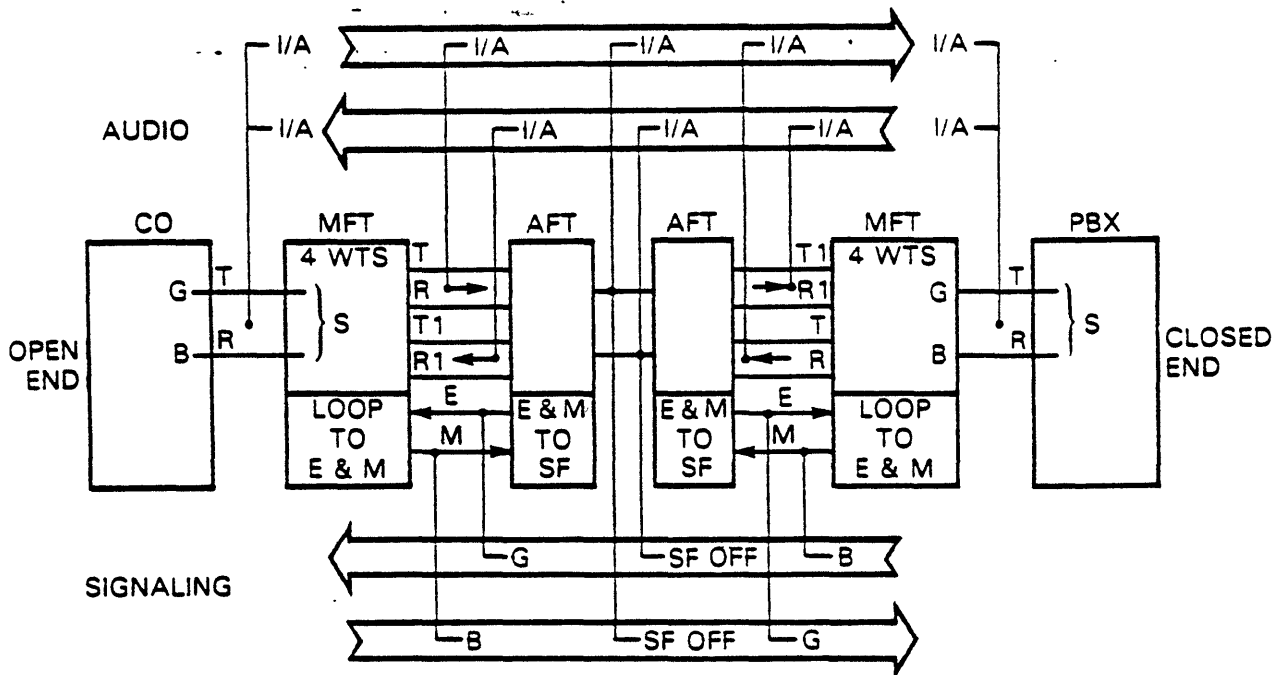


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**Figure 4-10e. Dial Pulse Addressing**

## Part I Trunking

**Occupied Circuit:** Once address signaling is complete and the distant end answers, an audio path is established in both directions and business may be conducted. (Figure 4-10f.)



10813A052

Figure 4-10f. Occupied Circuit

**INCOMING CALL (CO ORIGATION):** An incoming call is illustrated in the following figures:

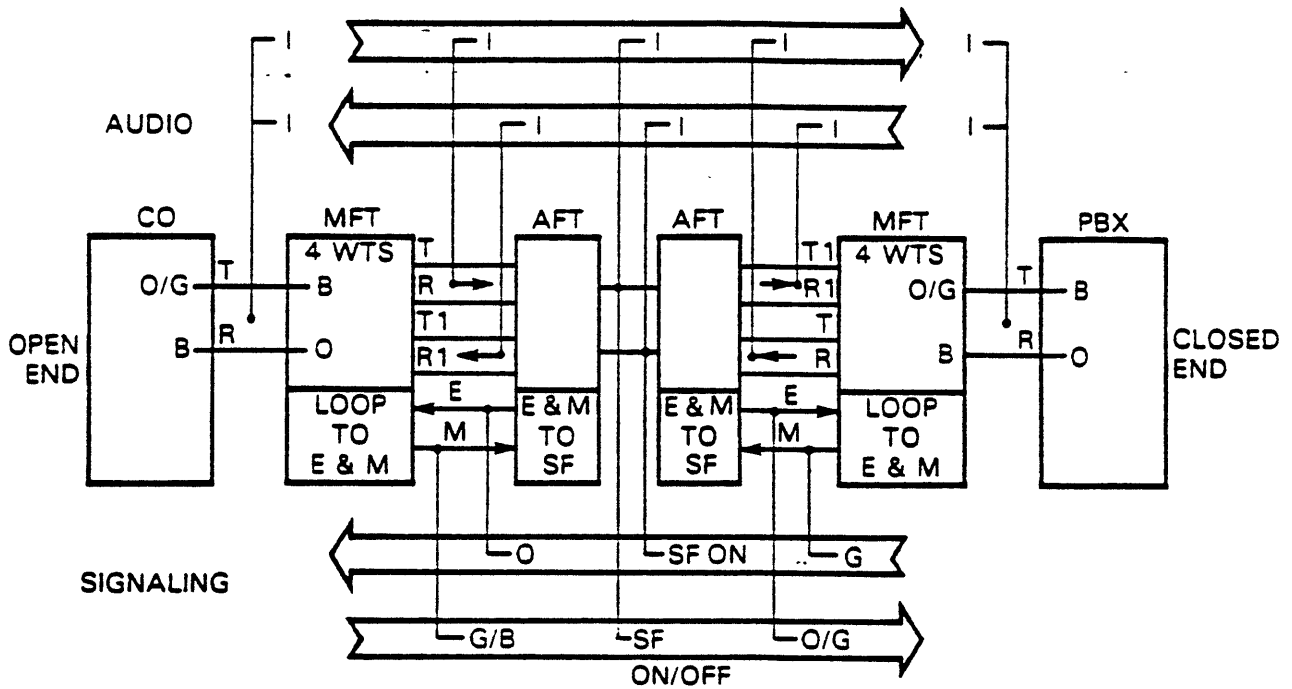
EVENT	FIGURE
CO Seizure	4-11a
Incoming Ring	4-11b
PBX Answer	4-11c
Occupied Circuit	4-11d

**CO Seizure:** The CO seizes the circuit by grounding the tip lead toward the MFT, as in Figure 4-11a. The M lead toward the AFT goes from ground to battery, SF is turned Off, and the E lead toward the MFT goes from open to ground. In the MFT, the tip lead toward the PBX goes from open to ground. The PBX recognizes this transition as incoming seizure and will not select the circuit for an outgoing call.

**Incoming Ring:** At the CO end, 20 Hz ringing voltage superimposed on battery is applied to the ring lead (2 seconds ring, 4 seconds rest). In wink-on-ring operation, the open-end MFT translates each ringing cycle into a momentary 'wink' that is signaled toward the closed-end MFT as in Figure 4-11b. The closed-end MFT contains a ringing supply and translates the 'wink' into a timed (2 seconds on, 4 seconds off) ring toward the PBX.

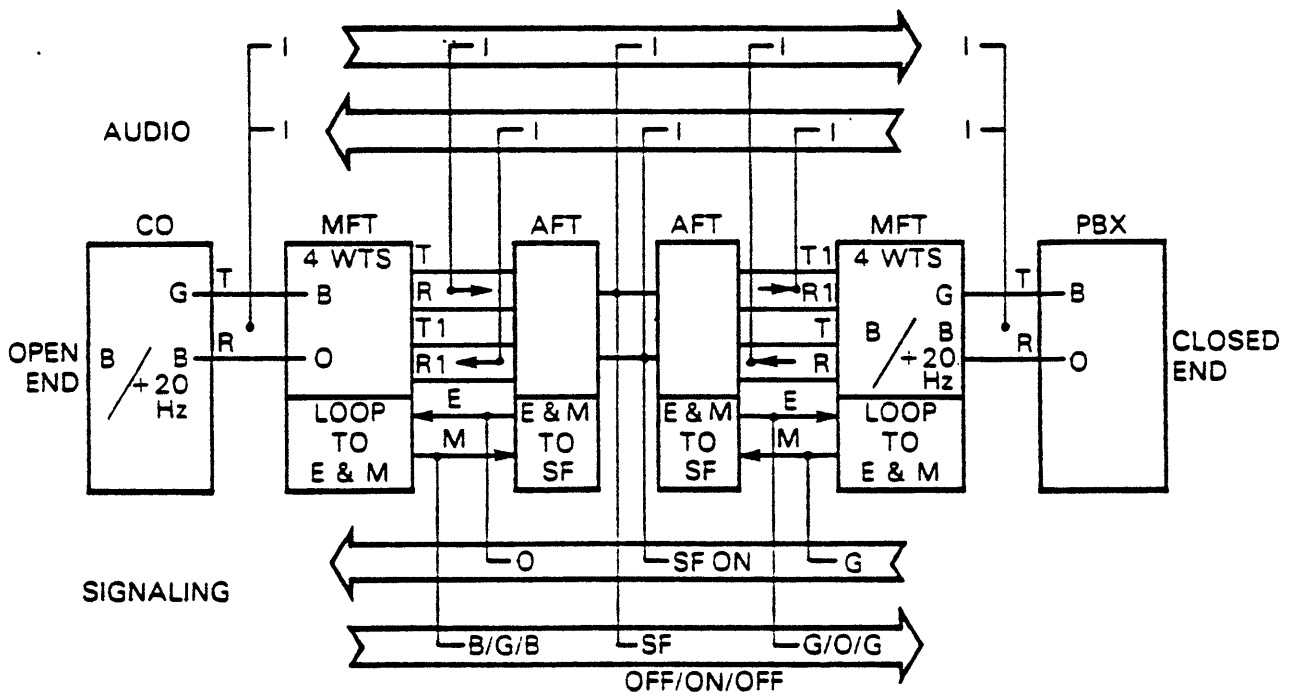
In ring-after-ring operation, ringing is signaled toward the closed end for the full ringing period. A specialized SF format where a 20 Hz signal is superimposed onto the SF tone is often used in the ring-after-ring operation.





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Figure 4-11a. CO Seizure



10813A054

Figure 4-11b. Incoming Ring

**Part I  
Trunking**

**PBX Answer:** The PBX answers by closing the loop between tip and ring. If the call is answered during the ringing period, dc flowing to the ringing supply over tip and ring will cause the ringing to cease immediately, preventing an annoying ring in the called party's ear.

The loop closure is signaled toward the open end as in Figure 4-11c.

Upon recognizing the loop closure in the open-end MFT, the CO removes the ringing supply and will not send another ringing cycle.

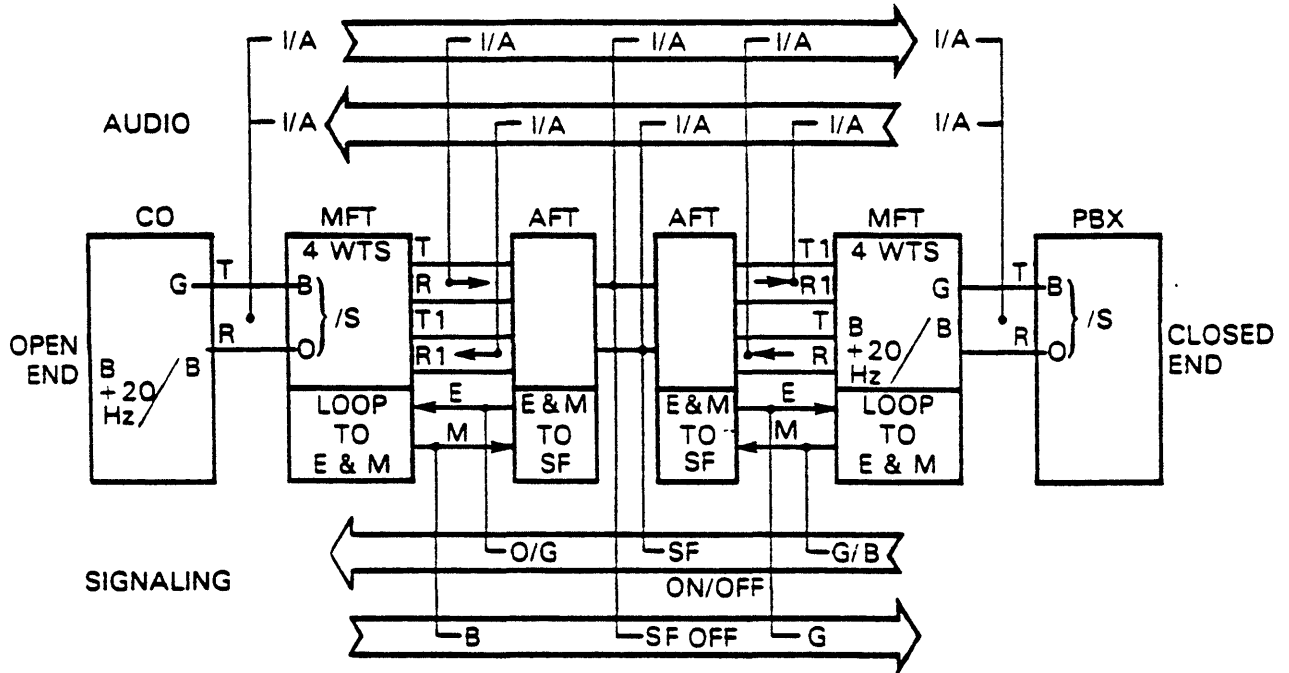


Figure 4-11c. PBX Answer

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**Occupied Circuit:** The circuit in the occupied condition is illustrated in Figure 4-11d.

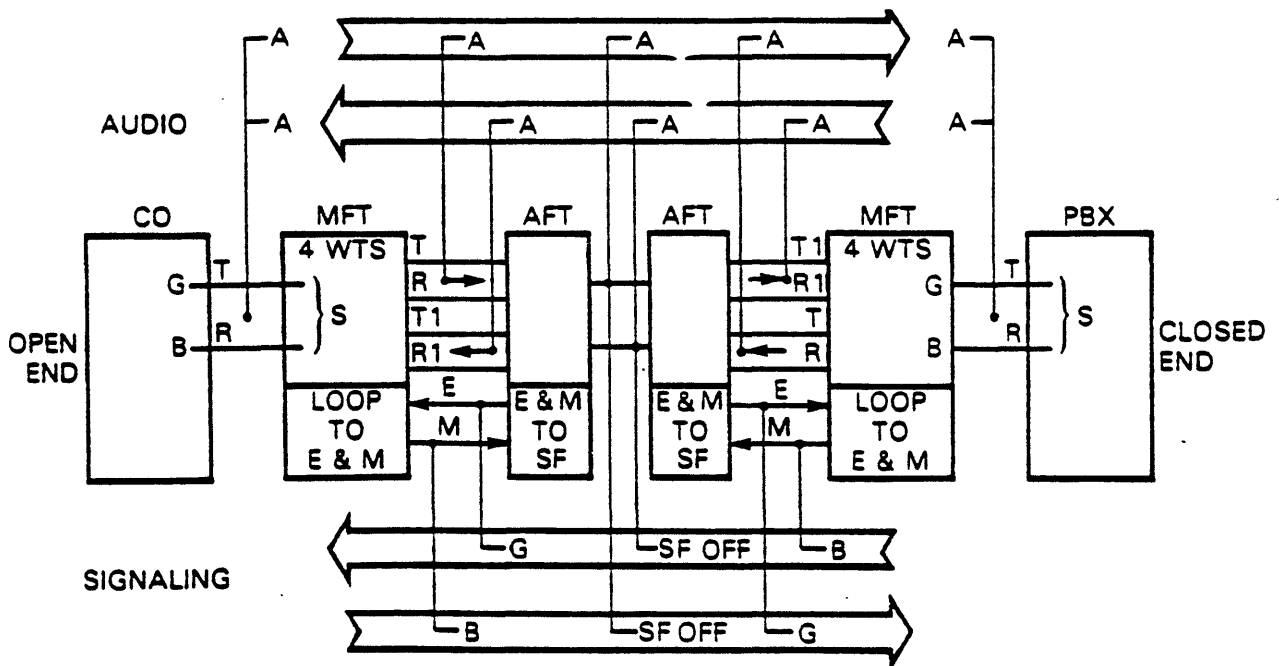


Figure 4-11d. Occupied Circuit

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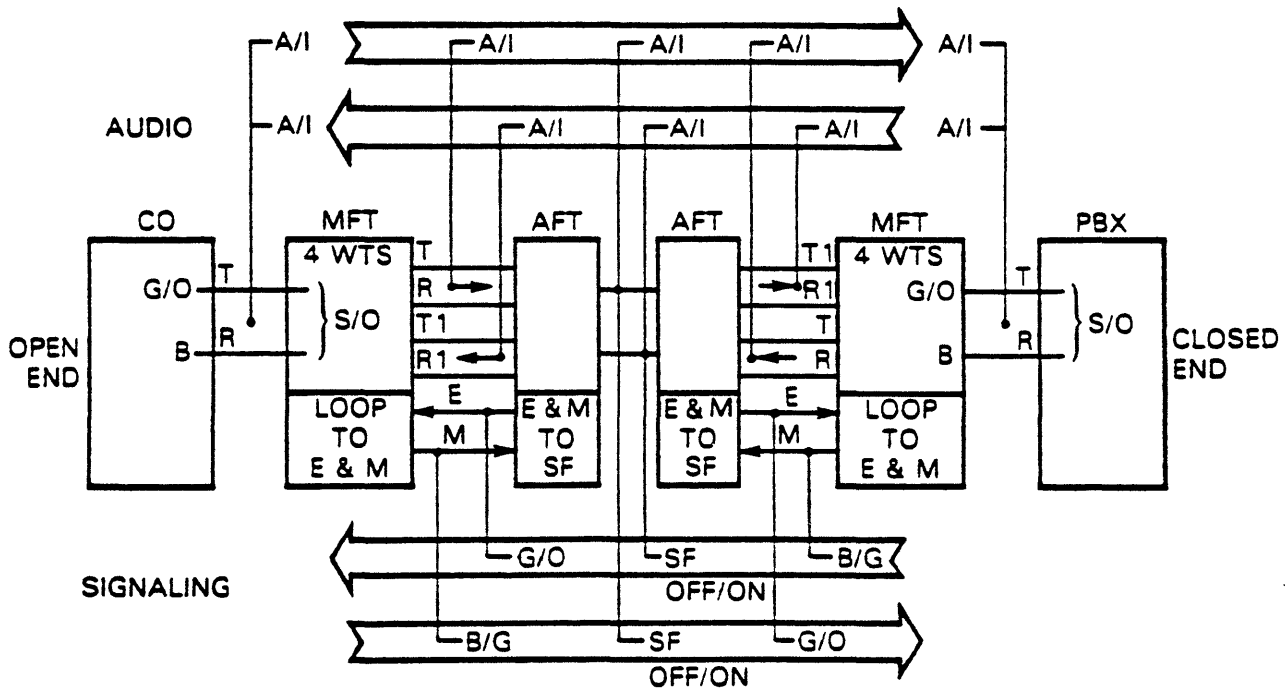
**CALL DISCONNECT:** Call disconnect is illustrated in the following figures:

EVENT	FIGURE
Disconnect	4-12a
Idle Circuit	4-12b

**Disconnect:** Disconnect may be initiated from either end.

At the CO end, disconnect is signaled by removing the ground from the tip lead. Going from left to right in Figure 4-12a, removal of tip ground translates into ground on the M lead toward the MFT. SF is turned On, and the E lead goes from ground to open at the closed-end MFT. In the closed-end MFT, ground is removed from the tip lead toward the PBX.

The PBX disconnects by opening the loop between tip and ring. The MFT translates open between tip and ring into ground on the M lead toward the AFT. SF is turned On, and the E lead toward the open-end MFT goes from ground to open. The loop closure toward the CO is removed.

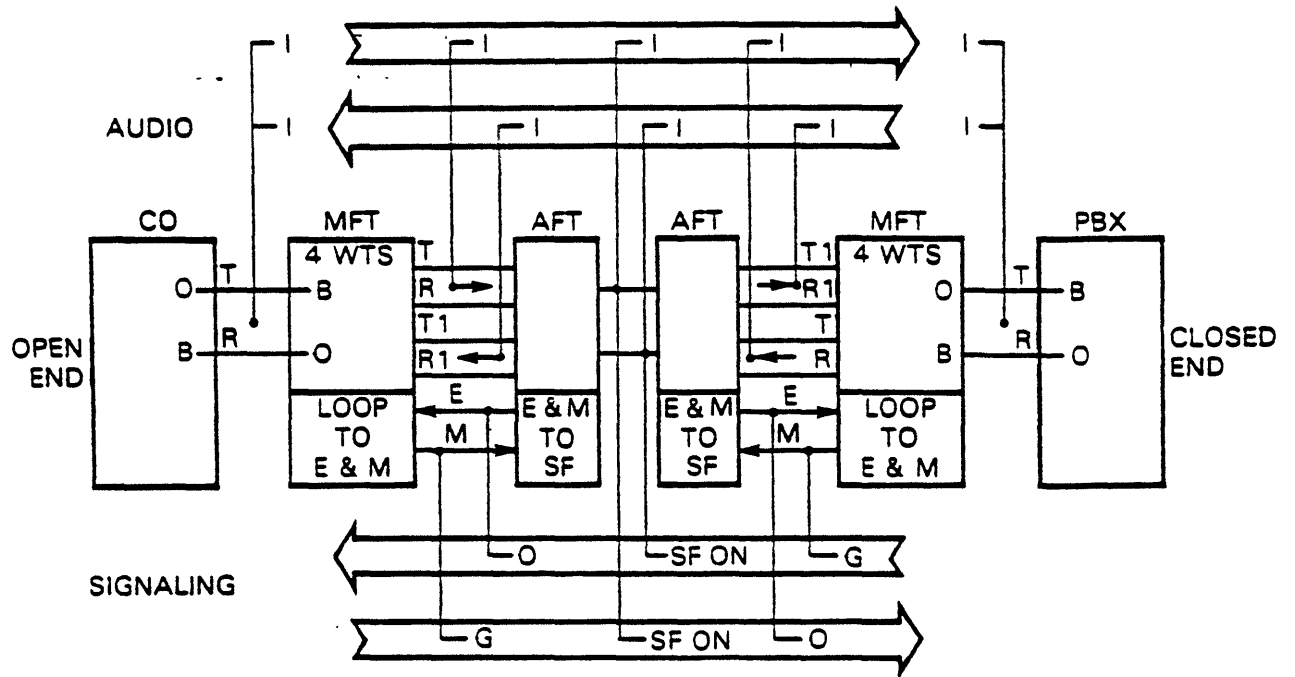


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**Figure 4-12a. Disconnect**

**Part I  
Trunking**

**Idle Circuit:** The circuit in the idle condition is illustrated in Figure 4-12b.



10813A058

**Figure 4-12b. Idle Circuit**

## **CHAPTER 4 REVIEW QUESTIONS**

1. What are the electrical states of the tip and ring leads in the idle condition for ground start:  
At the CO?  
At the PBX?
2. How does the CO respond to seizure (grounded ring lead) from the PBX?
3. Why is ground start signaling superior to loop start for senderized operation?
4. How does the CO originate a call?
5. Why is ground start signaling superior to loop start with respect to CO origination?
6. How does the CO respond to distant-end disconnect?
7. How is ringing signaled toward the closed end in wink-on-ring E&M lead operation?
8. How is ringing signaled toward the closed end in ring-after-ring operation?



## CHAPTER 5

# CONTROLLED OUTPULSING—DELAY DIALING AND WINK START OPERATION

### INTRODUCTION

To receive address information (dial pulse, DTMF or MF), a switching machine must be equipped with the appropriate digit detection and collection equipment. The most straightforward method of providing this equipment is to connect digit collection equipment to each trunk circuit. However, as we've seen elsewhere, substantial economics may be realized through the sharing of equipment.

Digit collection equipment must be present for only a small fraction of the duration of a typical call; therefore, the equipment may be shared among a number of individual circuits. Equipment that is shared in this manner is called *common equipment*.

When digit collection equipment is shared, terminating equipment must have some method of signaling originating equipment when digit collection equipment has been attached to a trunk.

As we have seen in non-senderized operation, dial tone is not transmitted to the originating end until digit collection equipment has been attached. In senderized loop start operation, outpulsing may be positively controlled only if the originating switch is capable of recognizing dial tone

In senderized ground start operation, the terminating switch does not ground the tip lead until it has attached digit collection equipment.

Four additional methods of controlled outpulsing have been developed for signaling between switching machines: *delay dialing*, *wink start*, *start dialing* and *stop-go*. Delay dialing and wink start operation will be discussed in this chapter.

Both delay dialing and wink start may be used on two-way trunks between switches. Since seizure is allowed from either end, glare becomes a consideration. As we will see, wink start provides better protection against glare than does ground start.

### DELAY DIALING

Delay dialing will be discussed with reference to the following figures:

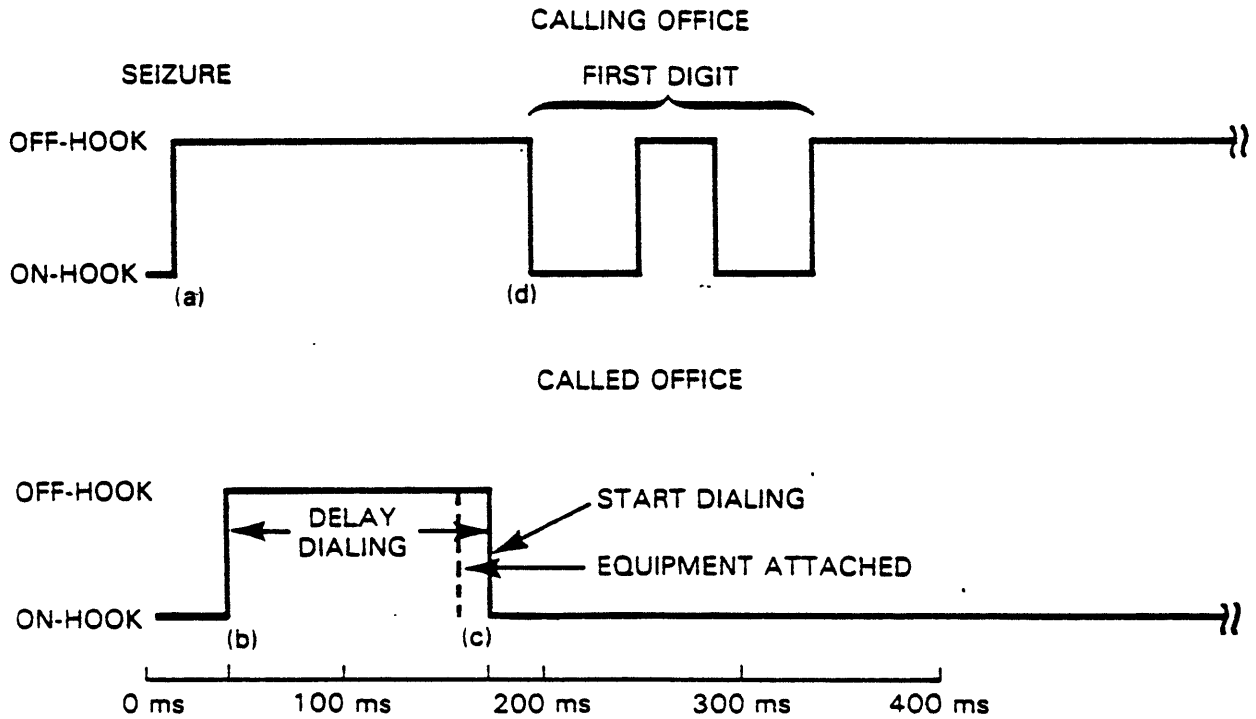
EVENT	FIGURE
Delay Dialing—Outpulsing Control	5-1 a
Delay Dialing—Entire Call	5-1 b

The figures are drawn to scale; however, the timing intervals are only approximate and may vary from application to application. The intent of the figures is to illustrate the sequence of events.

## Part I Trunking

In Figure 5-1a, the call begins when the calling office seizes the trunk toward the called office (a). The called office responds to the seizure by going off-hook (b). The called office remains off-hook until it is ready to receive address information; it then goes on-hook (c). The off-hook interval is the delay dialing signal.

Upon receiving the start dialing signal from the called office (c), the calling office proceeds to send address information via dial pulsing or DTMF addressing. Dial pulsing and the sequence of events through the first dialed digit are illustrated in Figure 5-1a.



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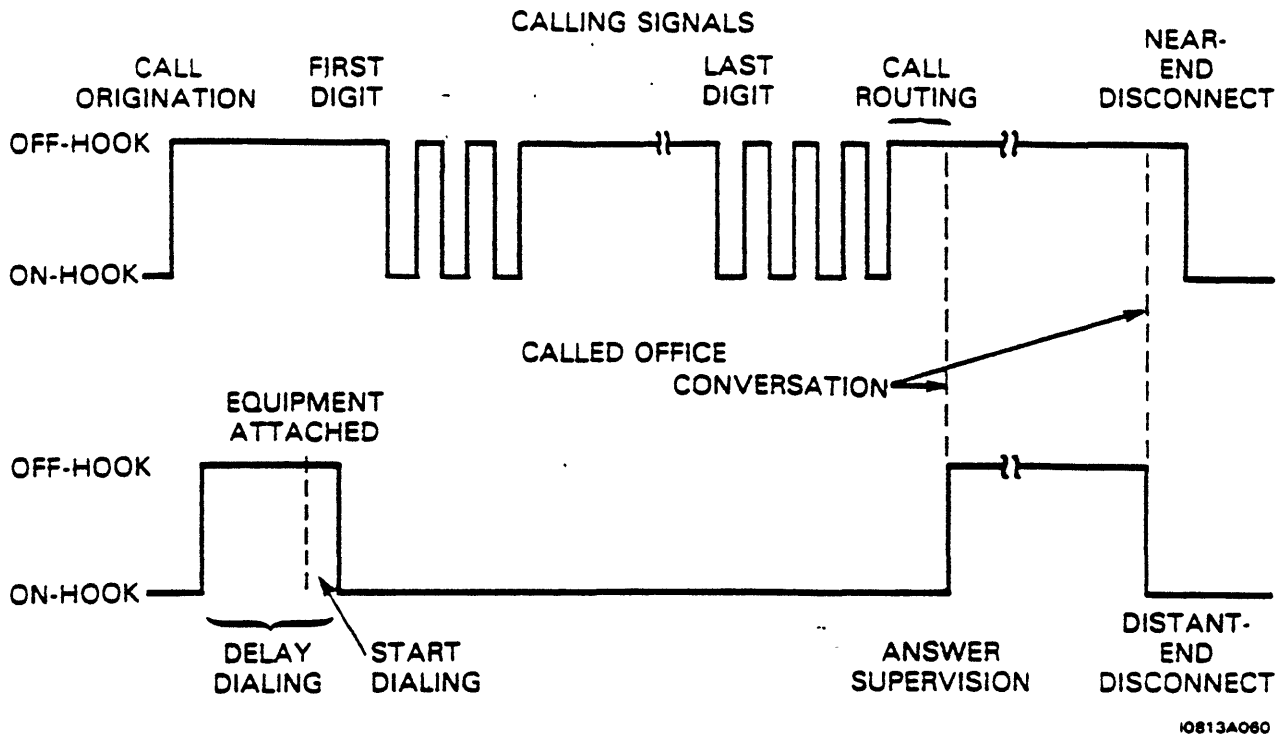
Figure 5-1a. Delay Dialing—Outpulsing Control

Figure 5-1b shows the entire call sequence, using a compressed time scale.

Once the calling office has transmitted all of the address digits, the call is routed to its destination. When the distant end answers, the called office signals answer supervision toward the calling office by going off-hook. Both offices remain off-hook for the duration of the call.

Either end may disconnect the call by going on-hook. Here the calling office disconnected first.





**Figure 5-1b. Delay Dialing—Entire Call**

**WINK START**

Wink start will be discussed with reference to the following figures:

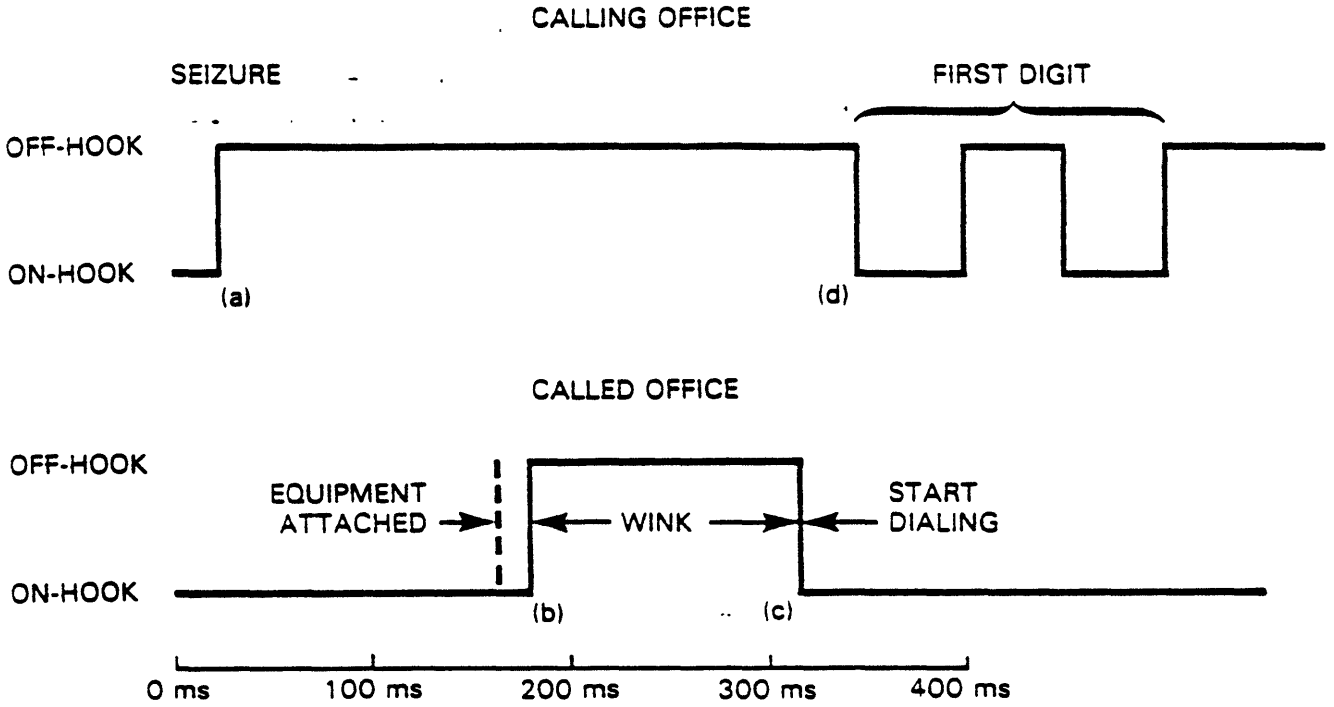
EVENT	FIGURE
Wink Start—Outpulsing Control	5-2a
Wink Start—Entire Call	5-2b

As with delay dialing, in wink start operation the calling office seizes the trunk by going off-hook (a) (see Figure 5-2a). Upon recognizing the seizure, the called office initiates a request for digit collection equipment; however, the called office does not immediately return an off-hook (delay dialing) signal to the calling office. The idle condition, on-hook signal is maintained until the digit collection equipment is attached at the called office.

Once equipment is attached, the called office goes off-hook (b) and maintains the off-hook for a specified interval.

The transition from on-hook to off-hook to on-hook constitutes the wink. Once the calling office receives the wink, it proceeds to send address information.

**Part I  
Trunking**



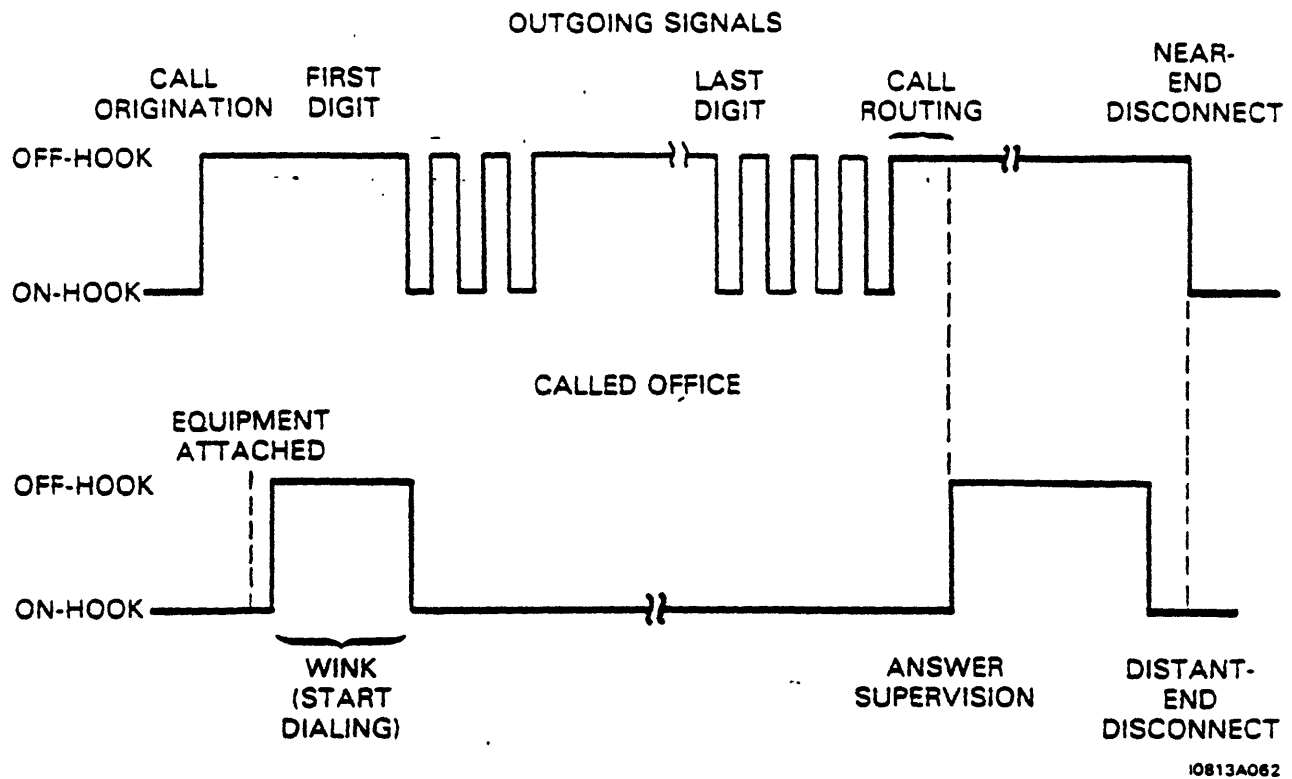
10813A061

**Figure 5-2a. Wink Start—Outpulsing Control**

Figure 5-2b shows the entire call sequence, using a compressed time scale.

Once the calling office has transmitted all of the address digits, the call is routed to its destination. When the distant end answers, the called office signals answer supervision toward the calling office by going off-hook. Both offices remain off-hook for the duration of the call.

Either end may disconnect the call by going on-hook. Here the called office disconnected first.



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Figure 5-2b. Wink Start—Entire Call

## GLARE

As has been discussed, two-way trunks are subject to occasional simultaneous seizure at both ends because of the unguarded interval between the seizure of the trunk at one end and the consequent busy condition of the trunk at the other end.

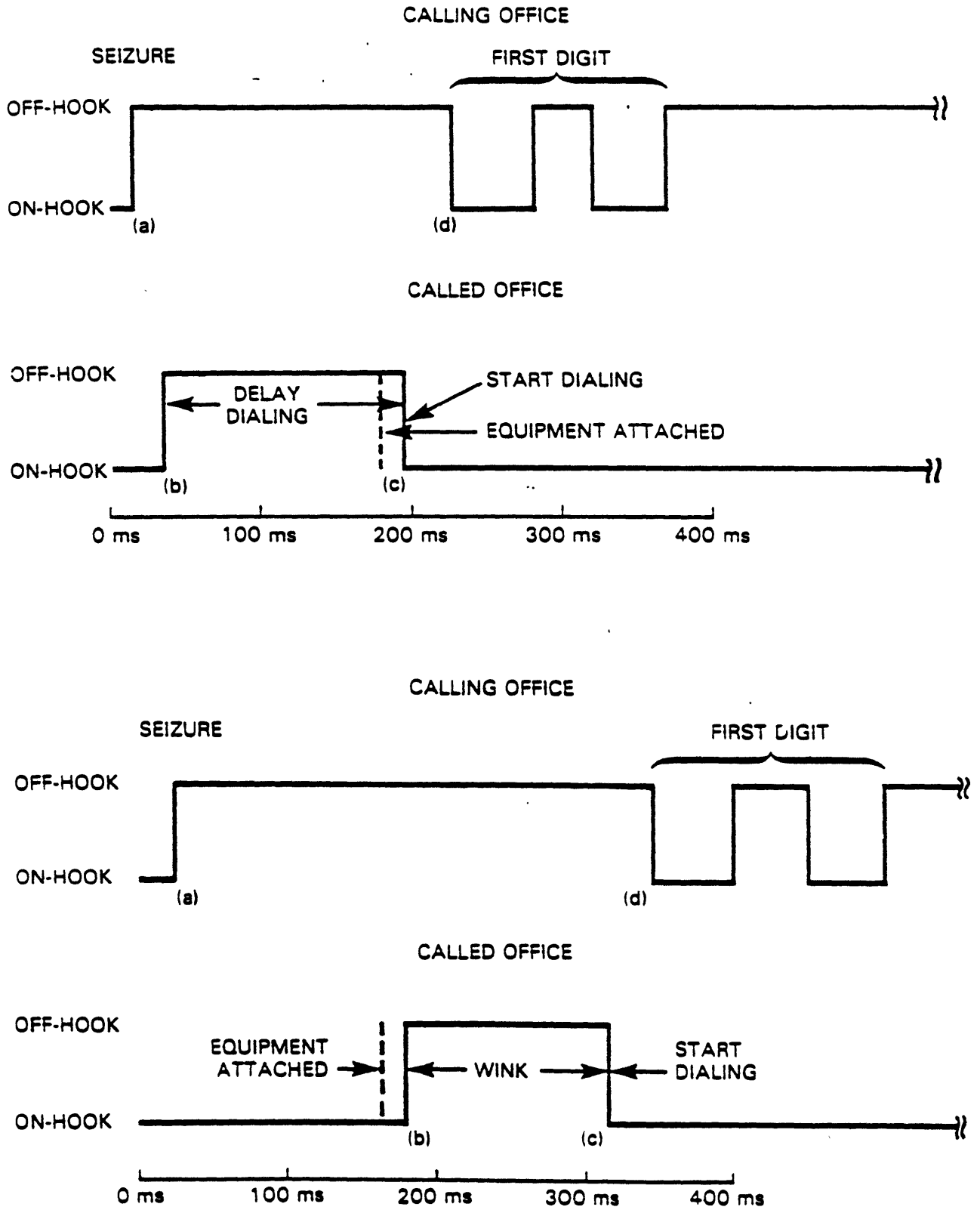
When delay dialing signals are used, the glare or blocked condition is assumed to exist if the off-hook condition persists for an interval greater than a specified time-out period. (Depending upon the application and equipment used, the time-out varies from 4 to 30 seconds.)

When wink start is used, the glare condition is assumed to exist if the off-hook interval is greater than the expected duration of the "wink." Wink start, therefore, provides faster recognition of the glare condition. This is shown in Figure 5-3.

In the case of both delay dialing and wink start signaling, a method must be provided to disengage from the mutually blocked condition.

Some types of common control equipment are arranged to flash forward at the reorder rate when the blocked condition is recognized. Upon receiving the flash, the distant end releases the connection and sends the reorder (fast busy or all trunks busy) signal to the call originator. The near end is also arranged to release the circuit and return reorder to the called party or, alternatively, to complete the call over the same trunk after a 2-second pause (if the start dialing or wink signal is received from the distant end).

**Part I  
Trunking**



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Figure 5-3. Comparison of Wink Start and Delay Dialing

## **CHAPTER 5 REVIEW QUESTIONS**

1. What is the purpose of controlled outpulsing?
2. In delay dialing, what is the start dialing signal?
3. In wink start, what is the start dialing signal?
4. Which method of controlled outpulsing provides better glare protection?



## CHAPTER 6 DIRECT INWARD DIALING

### INTRODUCTION

Direct Inward Dialing (DID) trunks allow an off-net caller to directly dial an extension served by a PBX, without the necessity of a call being served by an attendant. All calls over DID trunks originate at the switched public network and terminate at a PBX; therefore, DID trunks are called *one-way incoming trunks*.

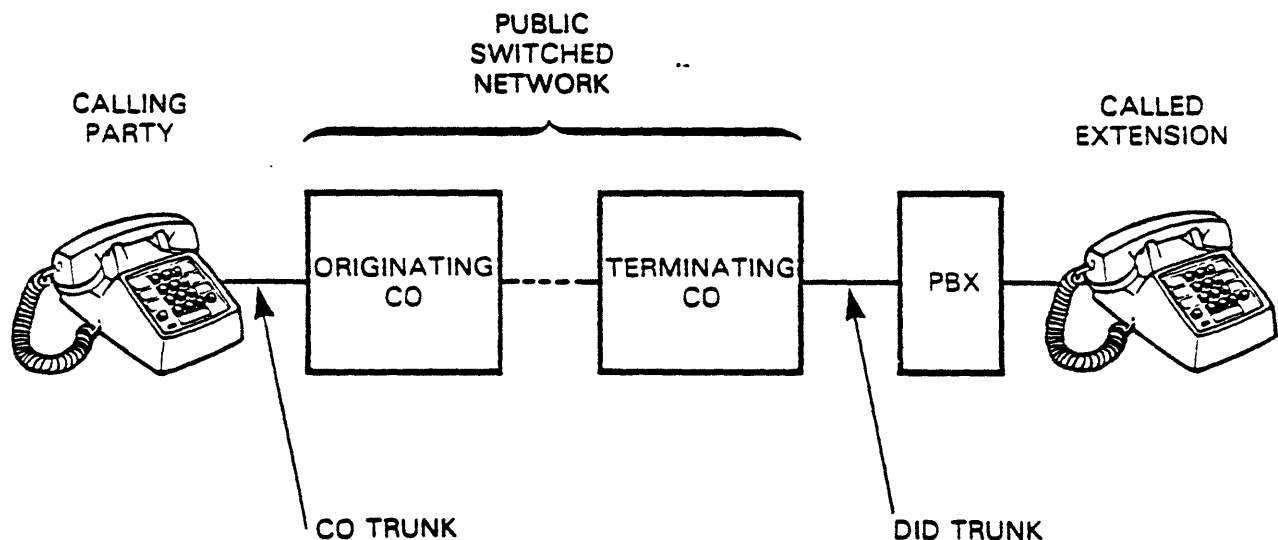
This chapter provides a functional overview of DID signaling followed by a detailed discussion of the signaling states in a DID call. Overall operation is discussed with reference to simplified circuit schematics.

### DID OVERVIEW

This section provides an overview of a DID call. The end-to-end connection is illustrated in Figure 6-1

A DID call is initiated when the off-net calling party goes off-hook and sends address digits to the originating CO. The call is routed through the public network to the terminating switch. Up to this point, the call is handled in the same manner as other calls placed over the public switched network.

The CO switch determines the trunk group (in this case, a DID trunk group) over which the call is to be completed to the customer's PBX. In addition, the CO switch determines the number of extension digits that are to be transmitted to the PBX. In general, the extension digits will consist of from one to four of the station digits (the station digits are the last four numbers dialed by the calling subscriber).



10813A064

Figure 6-1. A Typical DID Call

## Part I Trunking

The CO seizes an idle circuit in the DID trunk group. The PBX detects the seizure and responds by attaching digit collection equipment to the circuit. When the PBX is ready to receive the extension digits, it generally sends a "wink" start dialing signal to the CO, which then sends the destination digit through dial pulse addressing.

The PBX collects the destination digits, routes the call to the appropriate line, and rings the extension. When the extension answers, the PBX sends an answer supervision signal to the CO and the connection between calling party and called party is complete.

Either party may disconnect the call by going on-hook.

### DETAILED OPERATION OF DID SIGNALING

The signaling states of DID operation will be discussed with reference to two viewpoints from the demarc: looking toward the CO and looking toward the PBX.

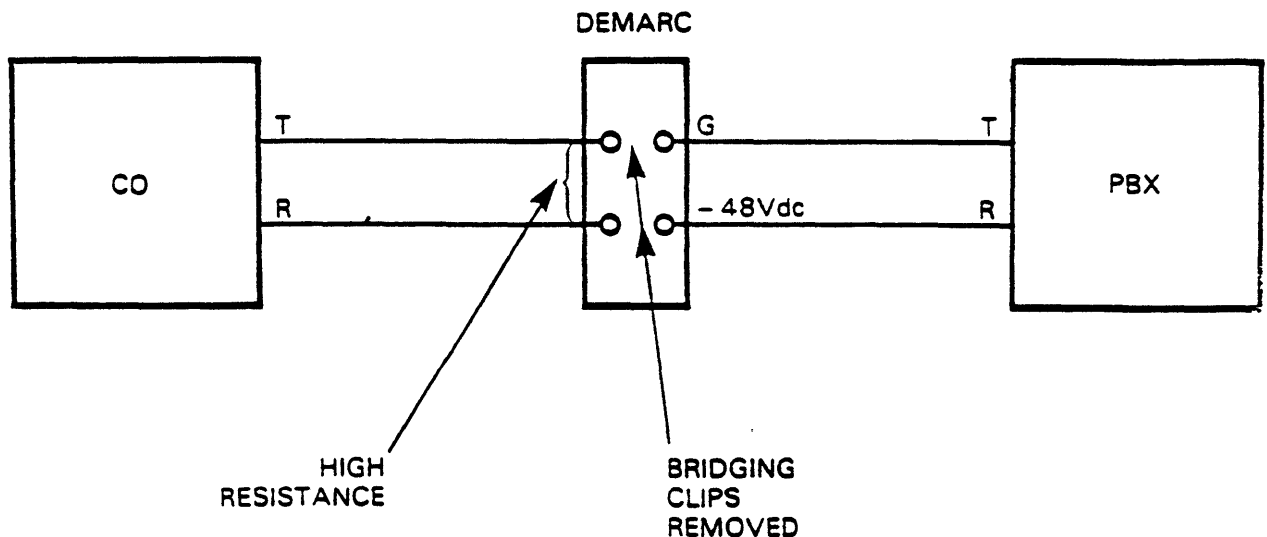
**IDLE CONDITION:** The idle condition is represented in Figure 6-2. Looking toward the PBX, ground is observed on the tip, and battery is observed on the ring. Looking toward the CO, a high resistance loop is observed between tip and ring.

**INCOMING CALL, CO END:** To test an incoming call toward the CO end, the bridging clips are removed and a voltage-ohmmeter (VOM), set to read ohms, is connected as in Figure 6-3.

An appropriate telephone number is dialed from a telephone connected to the public switched network.

When the CO seizes the circuit toward the demarc, the resistance between the tip and ring leads are observed to go from high to low.

When the CO goes on-hook, the resistance between the tip and ring leads are observed to go from low to high.



0813A065

Figure 6-2. Idle Condition



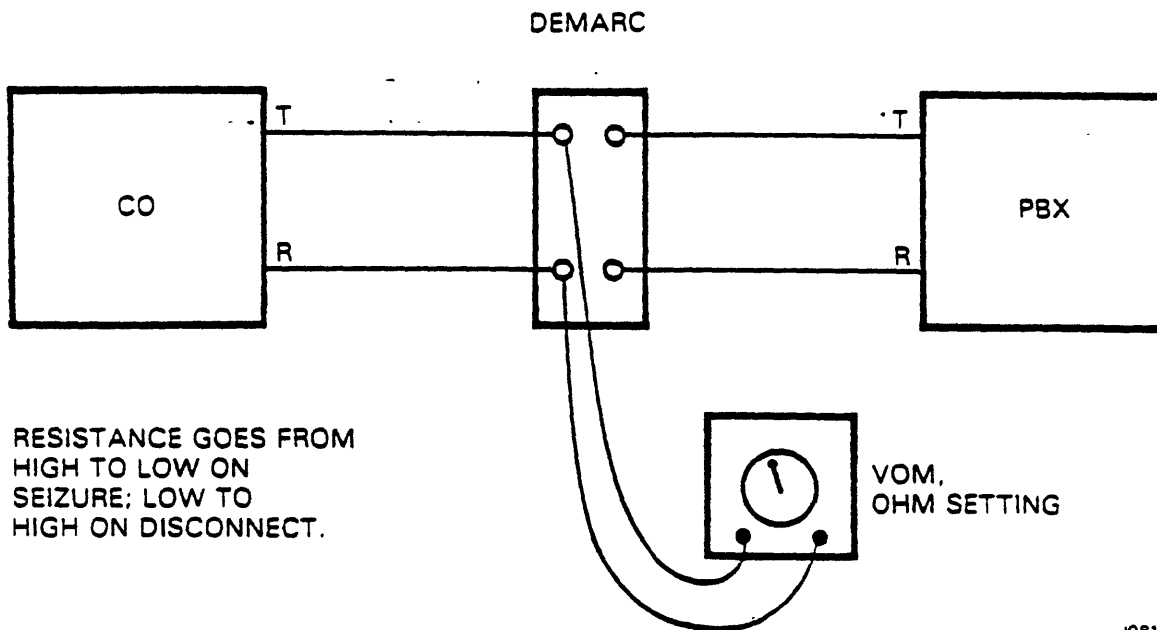


Figure 6-3. CO Seizure and Disconnect

**INCOMING CALL, PBX END:** To test an incoming call toward the PBX end, the bridging clips are installed and a voltmeter is connected as shown in Figure 6-4.

The number is dialed from a telephone connected to the public switched network.

When the CO seizes the circuit, a voltage drop on the ring lead is observed on the voltmeter because of the transition from high to low resistance.

If operating in the senderized mode, the start dialing signal from the PBX is observed as a brief reversal in the voltage observed at the demarc. This is the "wink." Otherwise, the CO will wait a predetermined amount of time before outpulsing.

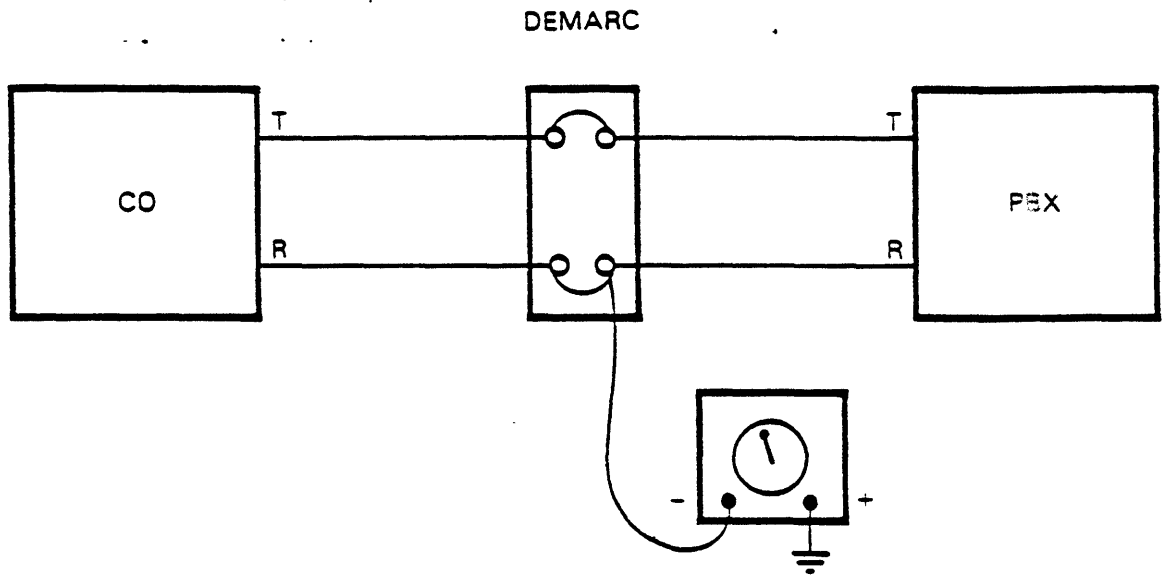
When the call is answered, the PBX places battery on the tip lead and ground on the ring lead. This is known as a T-R reversal. The voltage reversal will be observed on the voltmeter. (Due to the reversal of battery and ground on the tip and ring leads, this type of signaling is called loop reverse battery.)

**CALL DISCONNECT:** If the CO disconnects first, a brief voltage increase is observed as the loop in the CO switch goes from low to high resistance. This is followed by a voltage reversal when the PBX goes on-hook.

If the PBX disconnects first, a voltage reversal is observed, followed by an increase in voltage when the CO goes on-hook and the CO loop goes from low to high resistance.

Several test calls should be made. After each test call, the bridging clips should be removed and the circuit tested to ensure that it has returned to the idle condition.

**Part I  
Trunking**



10813A067

**Figure 6-4. PBX Operational Check**

## **CHAPTER 6 REVIEW QUESTIONS**

1. What is the purpose of Direct Inward Dialing?
2. What are the electrical states of the tip and ring leads in the idle condition:
  - In the PBX?
  - In the CO?
3. Can outgoing calls be made over DID trunks?
4. How does the CO seize the circuit?
5. What is the PBX start dialing signal toward the CO?
6. How does the PBX signal answer supervision to the CO?








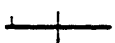

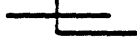
## APPENDIX A OVERALL LOOP START OPERATION

### OPERATION OF LOOP START SIGNALING

Overall operation of loop start signaling will be discussed with reference to the simplified circuit schematics in Figure A-1, Figures A-2a through A-2f, Figures A-3a through A-3c, and Figures A-4a through A-4c.

The symbols and conventions used in the figures are shown in Table A-1.

**Table A-1. Symbols and Conventions**

SYMBOL	EXPLANATION
	Heavy line. Indicates active portion(s) of the circuit for the operation being described.
	Light line. Indicates inactive circuitry for the operation being described.
	Light relay symbol. No current in relay windings, relay released.
	Dark relay symbol. Current in relay windings, relay operated.
	Relay in transition.
	Relay contact closed.
	Relay contact open.
	Audio path.

## Part I Trunking

**IDLE CONDITION:** Figure A-1 shows the circuit in the idle condition. All relays are in the released state and all relay contacts are as shown. Toward the CO, the tip lead is connected to ground through the L relay. The ring lead is connected to battery, also through the L relay. The relay is connected so that it will operate when it senses current flowing between battery on one side and ground on the other.

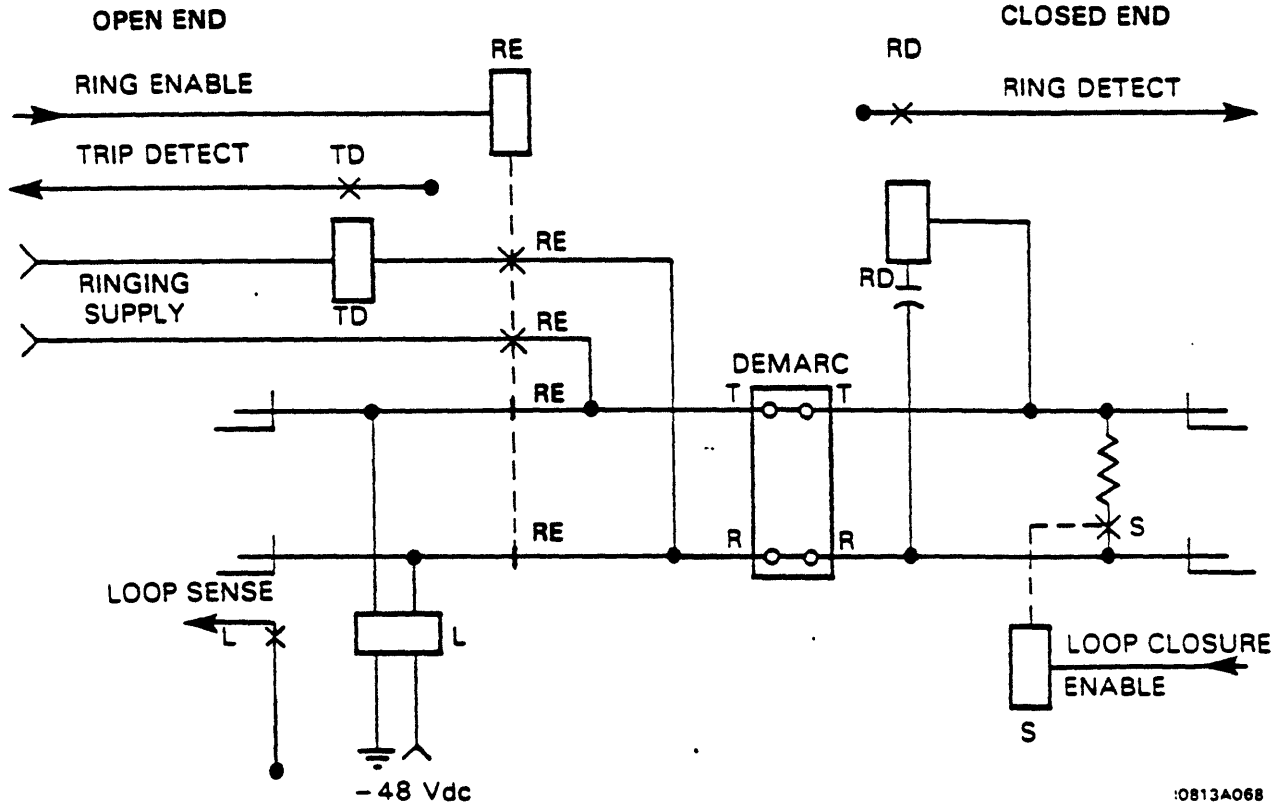


Figure A-1. Idle Condition

10813A068

**OUTGOING CALL (PBX ORIENTATION):** A call originated by the PBX will be discussed with reference to the events that take place between PBX origination (off-hook seizure) and call completion. The events and their associated circuit schematics are as follows:

EVENT	FIGURE
PBX Seizure	A-2a
CO Response	A-2b
DTMF Addressing	A-2c
Dial Pulse Break	A-2d
Dial Pulse Make	A-2e
Occupied Circuit	A-2f

**PBX Seizure:** The PBX seizes the trunk by operating the S relay via the loop closure enable lead. The S relay contacts close, causing current to flow in the loop between the PBX and the CO (Figure A-2a)

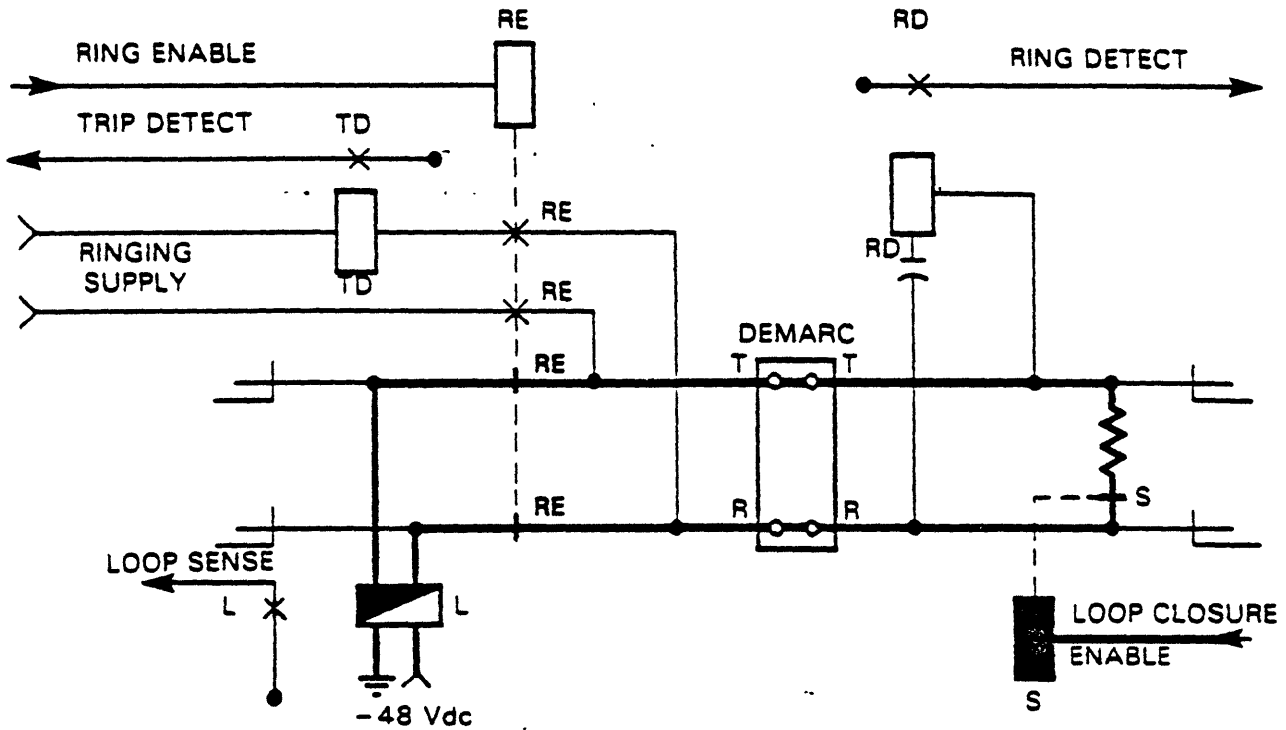


Figure A-2a. PBX Seizure

10813A069

**CO Response:** In the CO, loop current operates the L (loop sense) relay, closing the L contacts. Switching equipment detects the closure of the L contacts over the loop sense lead and attaches a dial tone generator and digit receiver to the circuit (Figure A-2b). Dial tone is sent to the PBX over the audio path.

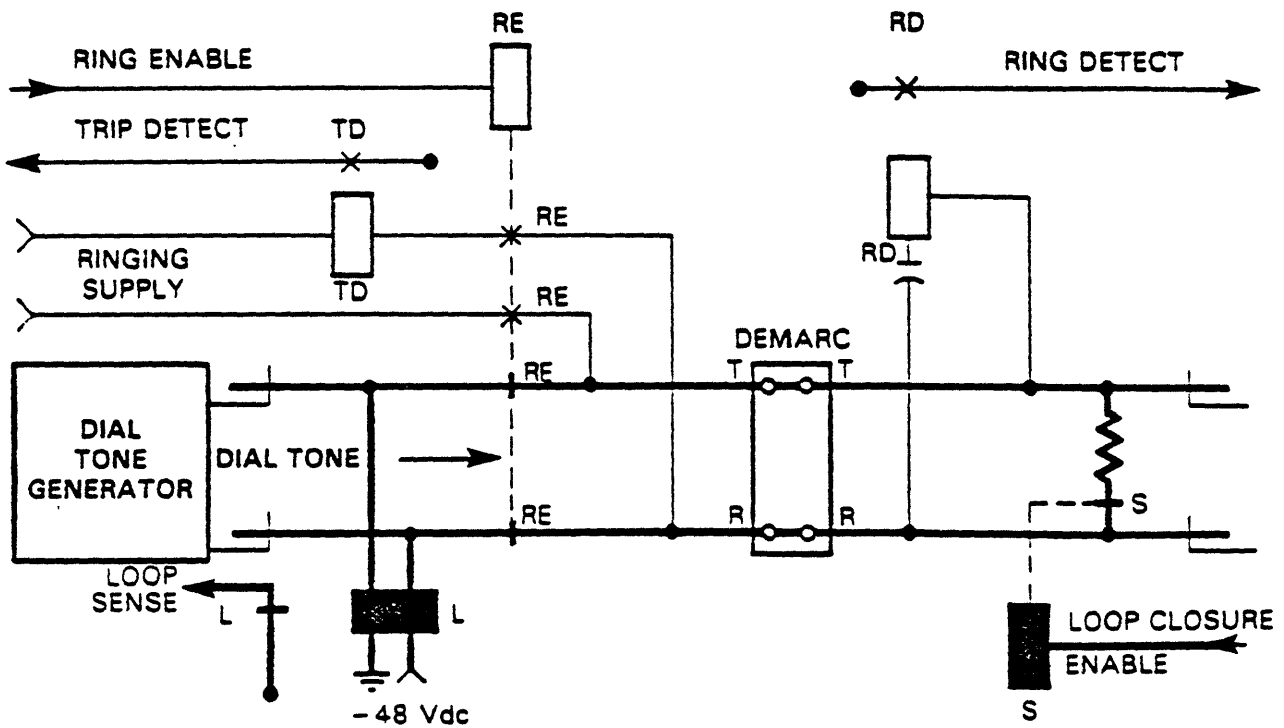


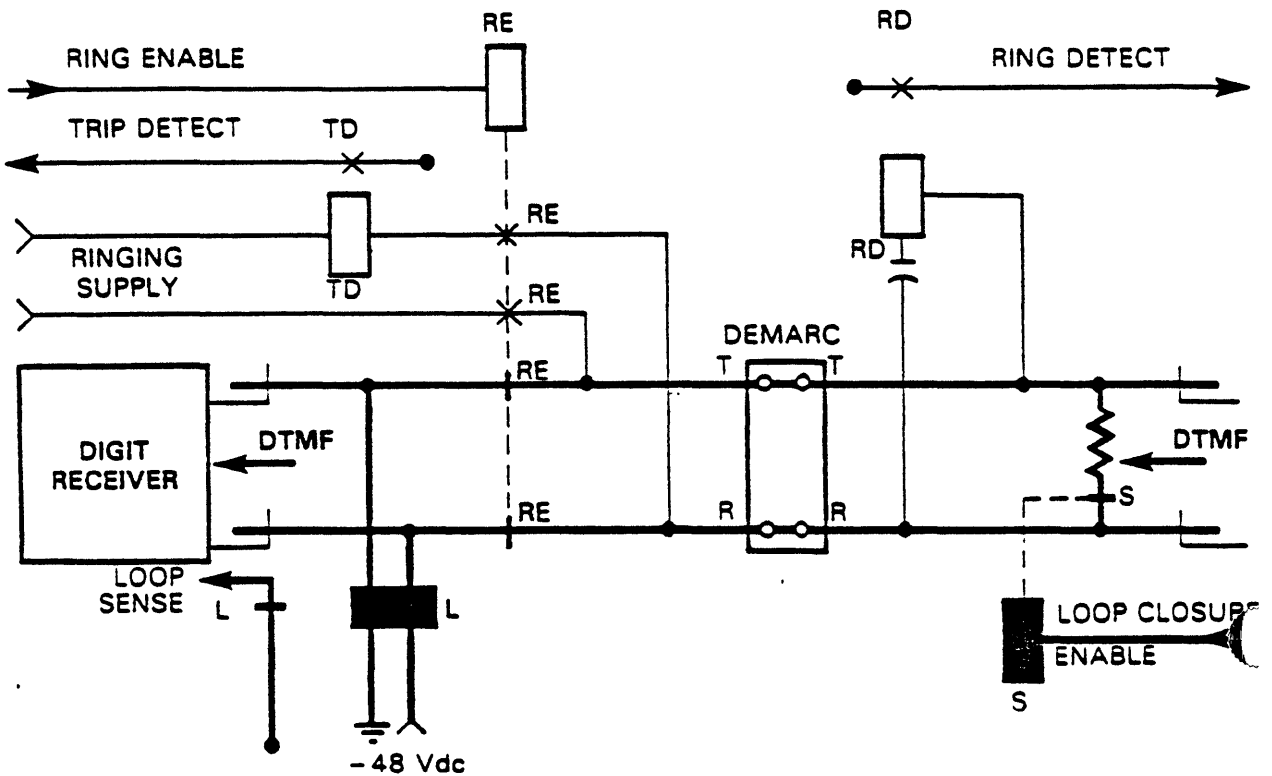
Figure A-2b. CO Response

10813A070

## Part I Trunking

**Address Signaling from PBX:** Upon receiving dial tone from the CO (or waiting a specified period after PBX goes off-hook, where dial tone detection equipment is not being used), the PBX begins outputting digits.

In the case of DTMF address signaling, DTMF digits are sent over the audio path and received by DTMF digit collection equipment connected to the audio path in the CO (Figure A-2c).



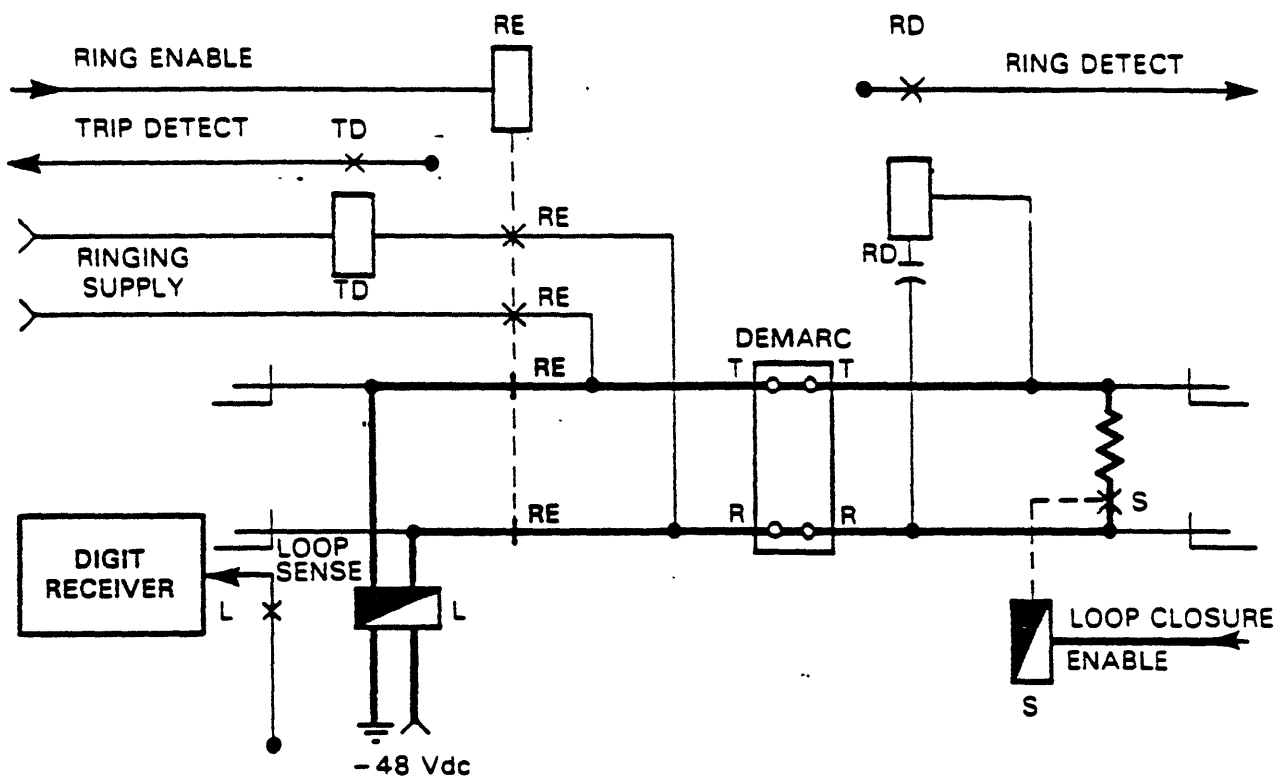
10813A071

Figure A-2c. DTMF Addressing

In dial pulse operation, dial pulses are sent by opening (break) (Figure A-2d) and closing (make) (Figure A-2e) the S contacts via the loop closure enable lead and the S relay. Each break releases the L relay, opening the L contacts; each make operates the L relay, closing the L contacts.

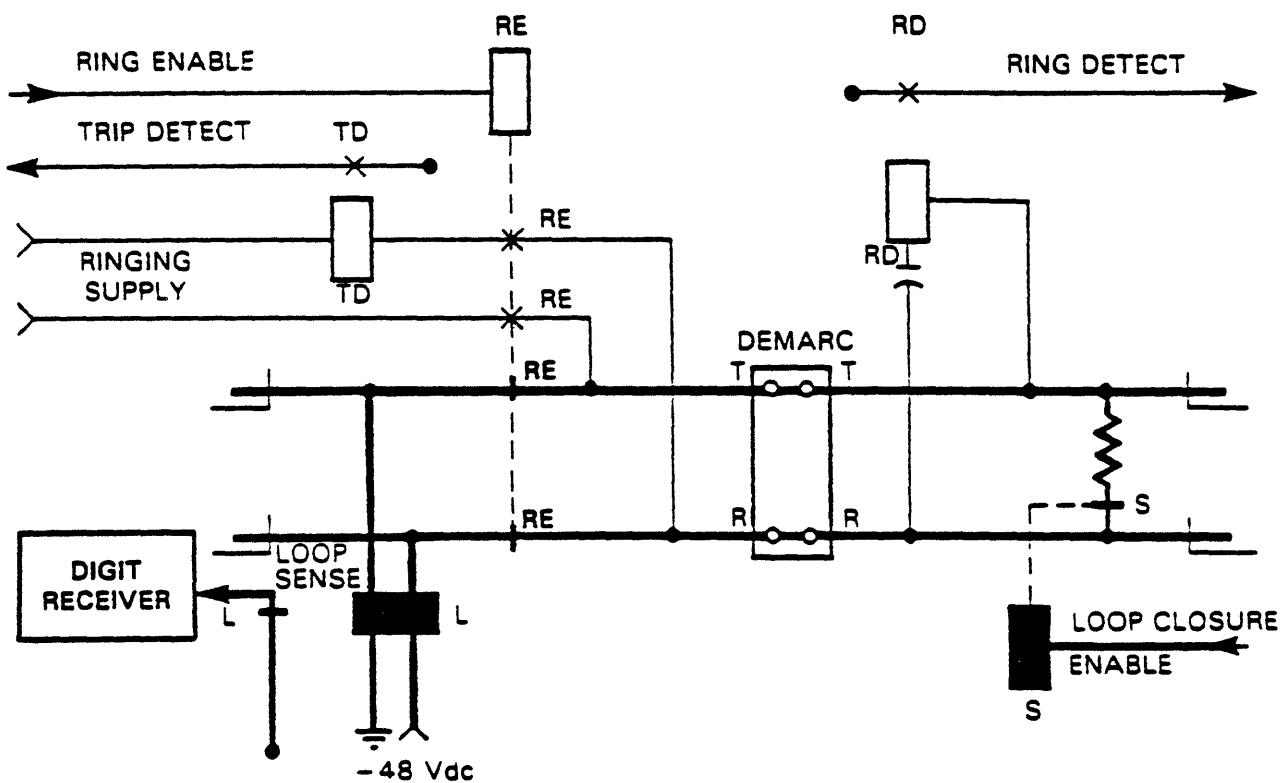
Dial pulses are transmitted to digit collection equipment over the loop sense lead. The equipment is able to tell the difference between the dial pulse break and an on-hook disconnect because they differ in duration. The dial pulse break is not mistaken for a disconnect because of its short duration.





10813A072

Figure A-2d. Dial Pulse Break



10813A073

Figure A-2e. Dial Pulse Make

## Part I Trunking

**Call Routing and Progress:** Once all digits have been collected, loop signaling between the CO and PBX is complete (except for call disconnect). The calling party receives call progress tones over the incoming audio path as the call is routed through the network toward its destination.

**Distant-end Answer:** An off-hook answer at the distant end completes the two-way audio path between the distant-end station equipment and the PBX. The connected call is illustrated in Figure A-2f.

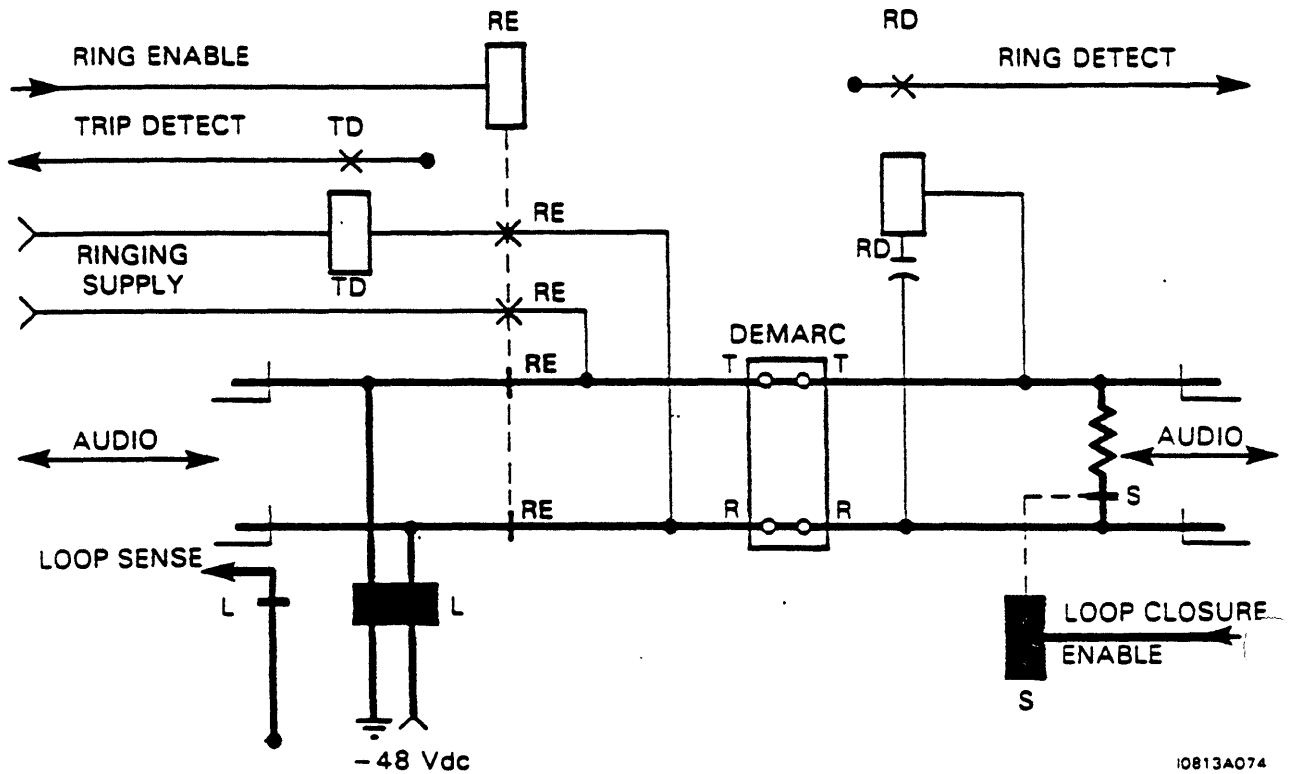


Figure A-2f. Occupied Circuit

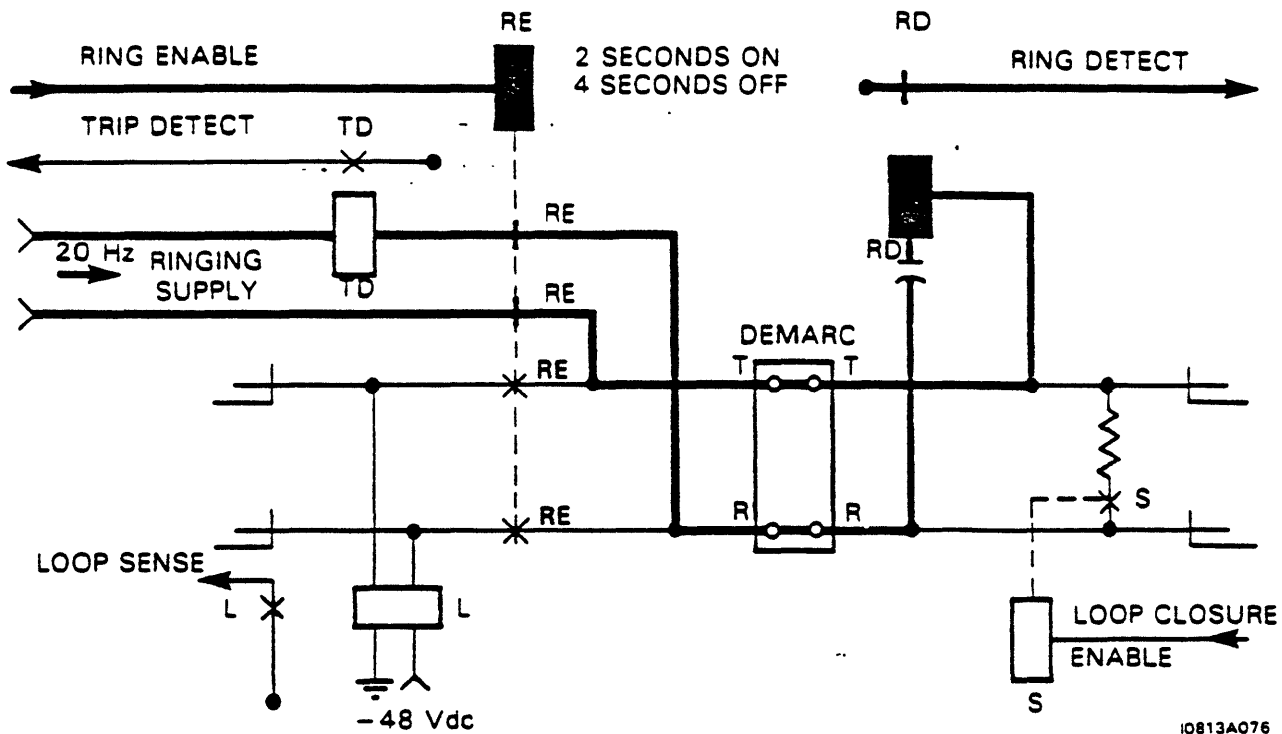
**INCOMING CALL (CO ORIGATION):** An incoming call will be discussed with reference to the events that take place between call initiation by the CO (incoming ring) and call completion. The events and their associated circuit schematics are as follows:

EVENT	FIGURE
Incoming Ring	A-3a
Ring Tripping During Ring	A-3b
Ring Tripping During Rest	A-3c

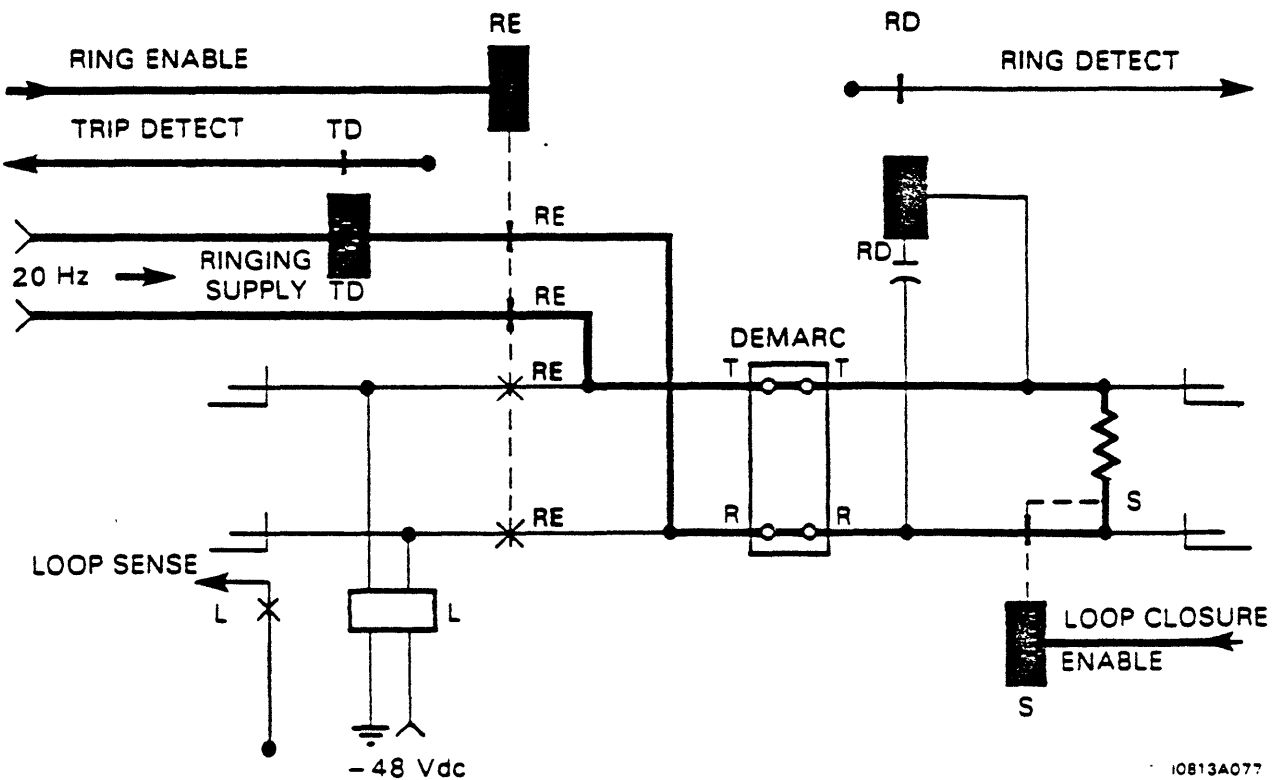
**Incoming Ring:** An incoming call to the PBX is initiated by the application of 20 Hz ringing voltage (75 to 110 Vac) to the ring lead via the ringing supply. Note that the ringing voltage is superimposed on the -48 Vdc on the ring lead. Ringing voltage is applied for approximately 2-second ringing periods followed by approximately 4-second rest periods. The tip lead remains connected to ground.

Ringing voltage operates the RD relay, which is sensitive to ac. The PBX detects incoming ring via the ring detect lead (Figure A-3a).

**PBX Answer:** Upon detecting incoming ring, the PBX operates the S relay via the loop closure enable lead. The S contacts are closed and current flows in the loop.

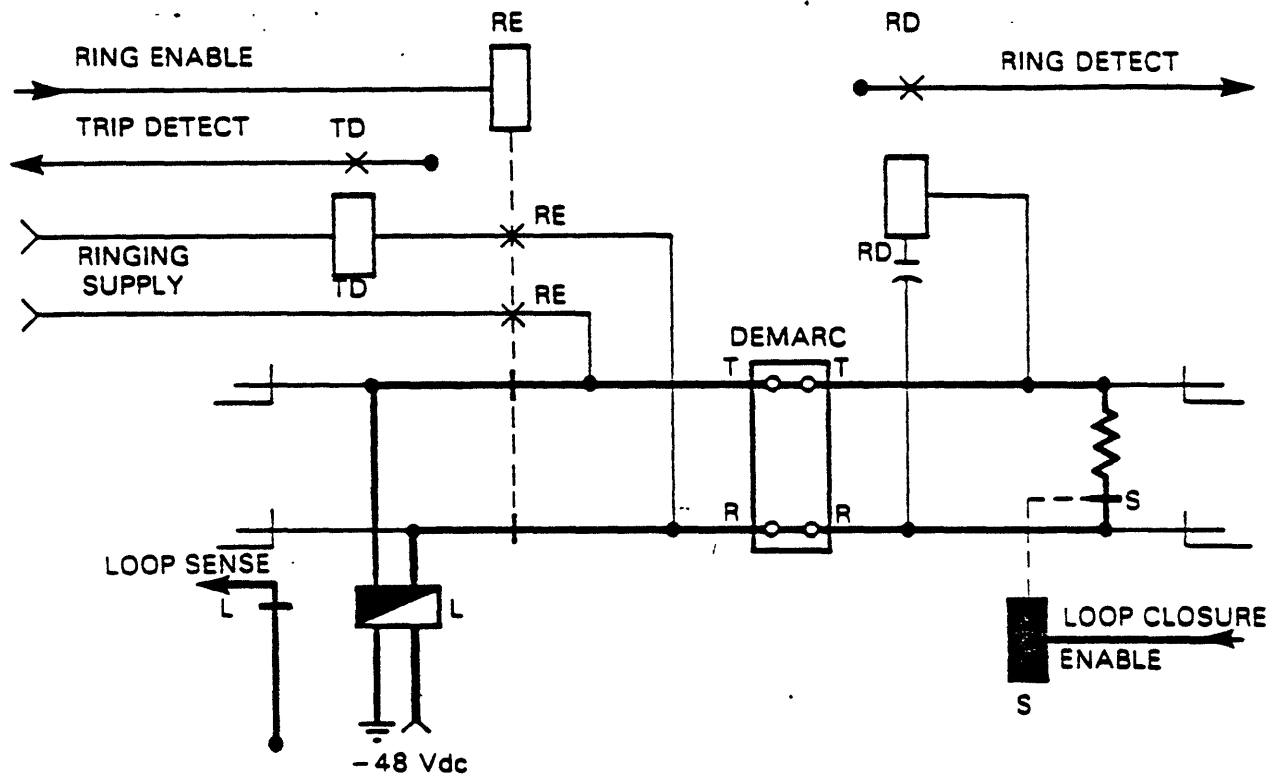


**Ring Tripping During Ring:** If the PBX closes the loop during the 2-second ringing period, dc loop current flows as shown in Figure A-3b. Loop current operates the TD relay, and the CO senses trip detect over the TD lead. The CO releases the RE relay via the ring enable lead. The RE contacts on the ringing supply are opened, and ringing ceases. The RE contacts on tip and ring are closed, completing the audio path and operating the L relay.



## Part I Trunking

**Ring Tripping During Rest:** During the rest period, the RE contacts on tip and ring are closed, as shown in Figure A-3c. Loop closure operates the L relay; the CO detects the answer over the loop sense lead and does not operate the RE relay to activate another ringing period.



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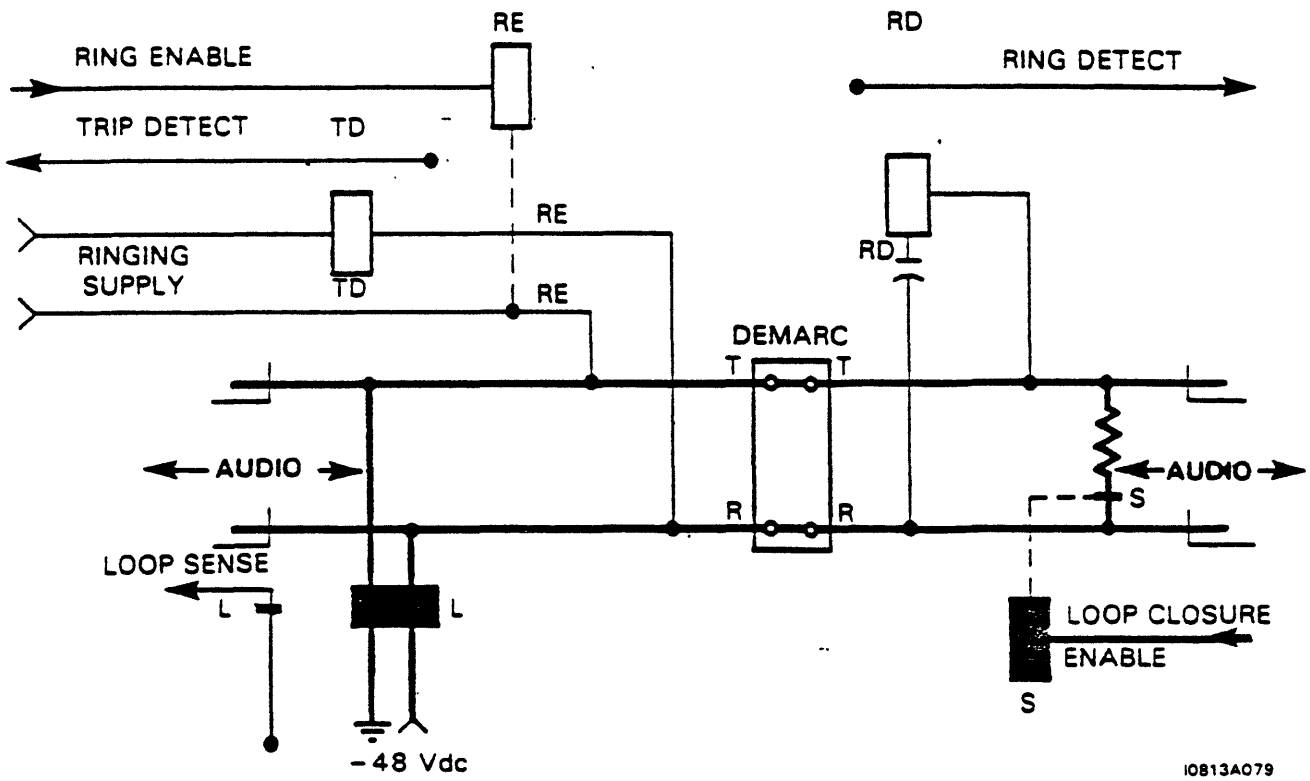
Figure A-3c. Ring Tripping During Rest

**CALL DISCONNECT:** Call disconnect is discussed with reference to Figures A-4a through A-4c. The events and associated figures are as follows:

EVENT	FIGURE
Occupied Circuit	A-4a
Near-end Disconnect	A-4b
Idle Circuit	A-4c

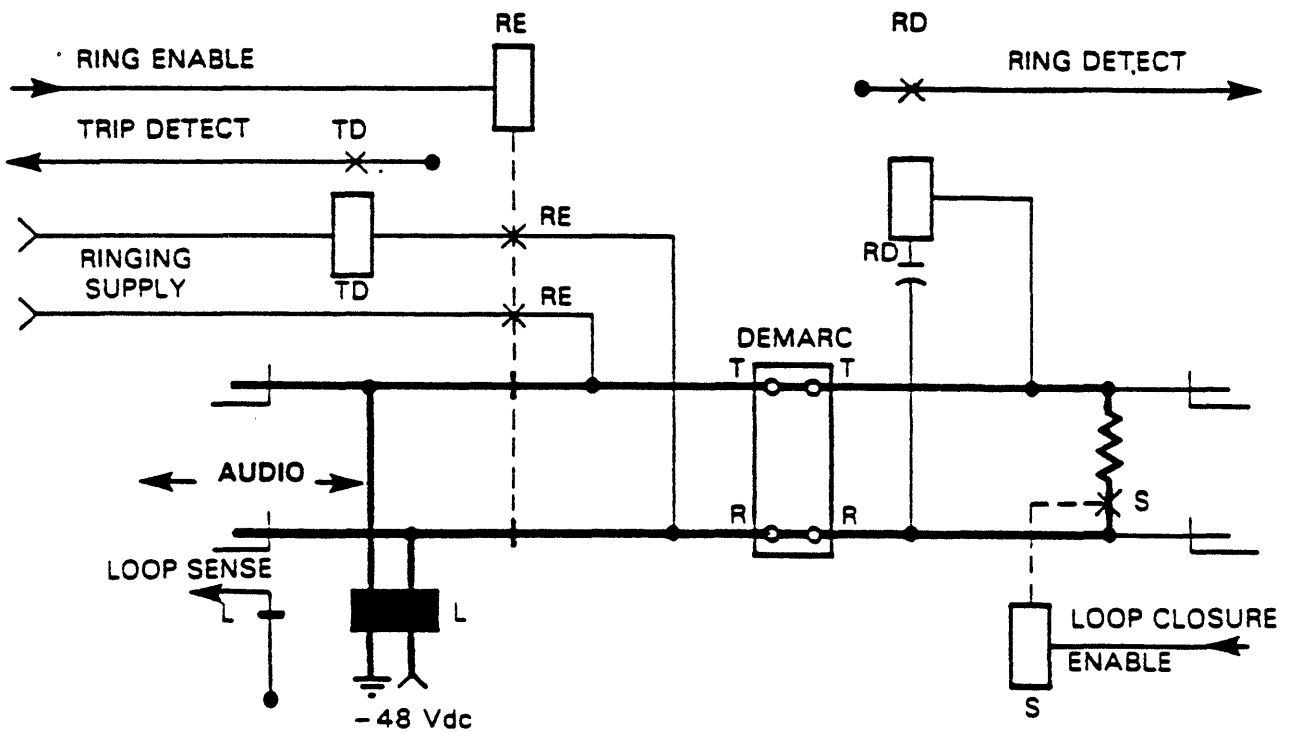
**Occupied Circuit:** The schematics associated with a connected call are shown in Figure A-4a.

**Near-end Disconnect:** A disconnect at the near end is initiated by the PBX station going on-hook. The PBX recognizes the on-hook and releases the S relay via the loop closure enable lead. The S contacts are open and current ceases in the loop (Figure A-4b).



10813A079

**Figure A-4a. Occupied Circuit**



10813A080

**Figure A-4b. Near-end Disconnect**

## Part I Trunking

**Idle Circuit:** In the CO, removal of current releases the L relay and the CO detects the on-hook via the loop sense lead (Figure A-4c).

CO circuitry is returned to the idle condition.

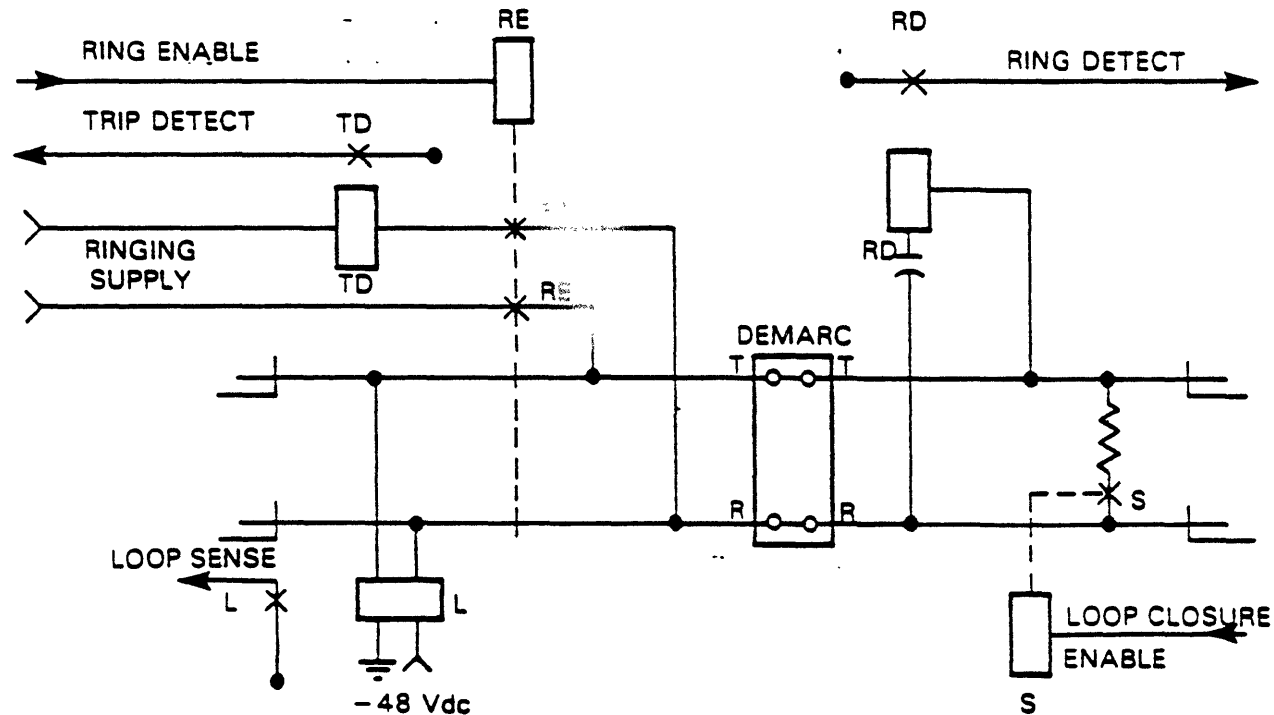


Figure A-4c. Idle Circuit

10813A081

**Distant-end Disconnect:** A distant-end disconnect is initiated when the distant-end station goes on-hook. The on-hook will be detected by the distant-end switch and the connection between the distant end and the CO will be "taken down."

The CO has no way of signaling the disconnect to the PBX; the call will not be disconnected until the near-end subscriber goes on-hook and initiates the near-end disconnect sequence.

### OPERATION OF LOOP START TO E&M INTERFACE

Overall operation of the loop start to E&M interface will be discussed with reference to Figures A-5a through A-5e (outgoing call) and Figures A-6a through A-6c (incoming call).

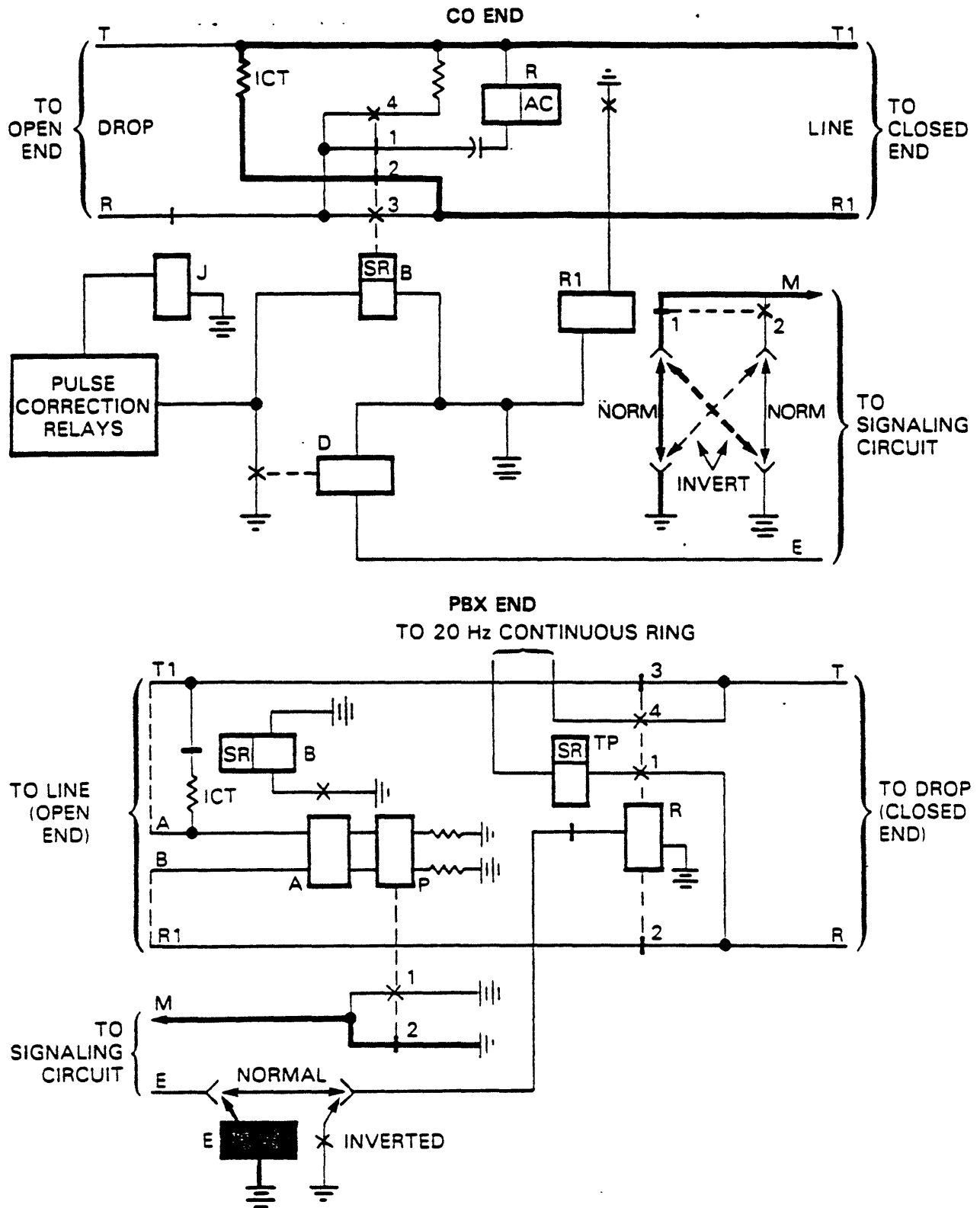
Each figure consists of two simplified circuit schematics—one for the CO end and one for the PBX end. In each figure, both normal and inverted E&M lead operation are shown. As stated previously, inverted operation should be used when SF is used as the facility signaling system

**OUTGOING CALL (PBX ORIGINATION):** An outgoing call is illustrated by breaking the call into the following events and their associated figures.

EVENT	FIGURE
Idle Condition	A-5a
PBX Seizure and CO Response	A-5b
Dial Pulse Break	A-5c
DTMF Addressing	A-5d
Occupied Circuit	A-5e

# Part I Trunking

**Idle Condition:** The idle condition is represented in Figure A-5a. Relays and the relay contacts are as shown. The signaling directions for the E&M leads are indicated by the arrows.



10813A082

Figure A-5a. Idle Condition

## **Part I Trunking**

**PBX Seizure and CO Response:** The PBX seizes the circuit by closing the loop between the tip and ring leads, as illustrated in Figure A-5b. Current in the loop operates the A relay; the B relay, which is a slow-release relay, follows. The B contacts open, opening the high-resistance idle circuit termination (ICT) bridge. Current in the loop also operates the P relay, which opens the P(2) contacts and closes the P(1) contacts. This changes the M lead from ground to battery, signaling the off-hook condition toward the signaling circuit and the facility. Incoming signaling on the E lead remains unchanged and is open for normal operation and ground for inverted operation. Note that the potential at point A is open for either normal or inverted operation.

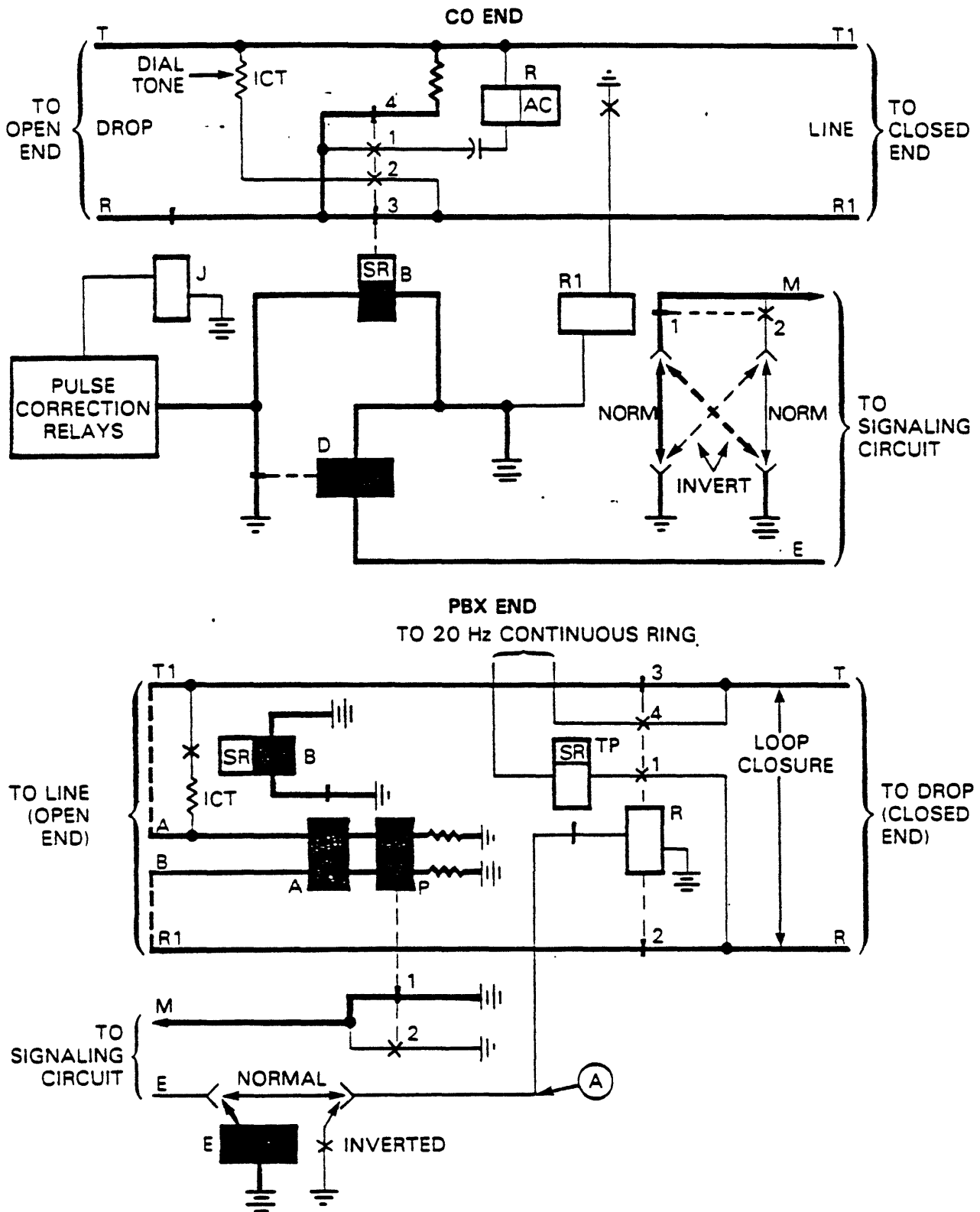
The CO, or open end, receives the off-hook as ground on the E lead. Current in the E lead operates the D relay, closing the D contact. The B relay follows. The B(2) contacts open, opening the ICT bridge. The B(1) contacts open, which ensures that the R relay will not operate during the call, preventing an erroneous ring from being signaled toward the PBX. B(4) closes, completing the loop between tip and ring, and B(3) closes, completing the audio path. Upon sensing loop current, the CO attaches a digit receiver and dial tone generator to the line and transmits dial tone toward the CO. The J relay does not operate because the pulse-correction relays allow current to flow toward the J relay only for short-duration dial pulse breaks.

Note that outgoing signaling on the M lead remains in the idle state—ground in normal operation and battery in inverted operation.

**Dial Pulse Break:** Momentary openings of the loop in the PBX cause the P relay to release, changing the state of the M lead. Although the A relay releases and opens the A contacts, the B relay does not release because it is a slow-release relay.

At the CO end, the E lead causes the D relay to release, opening the D contact. The pulse correction relays respond to the dial pulse breaks, and the J relay operates, opening the loop at the J contacts. The B relay does not release, as it is a slow-release relay. Upon receiving the first digit, the CO removes the dial tone from the line. (Figure A-5c.)

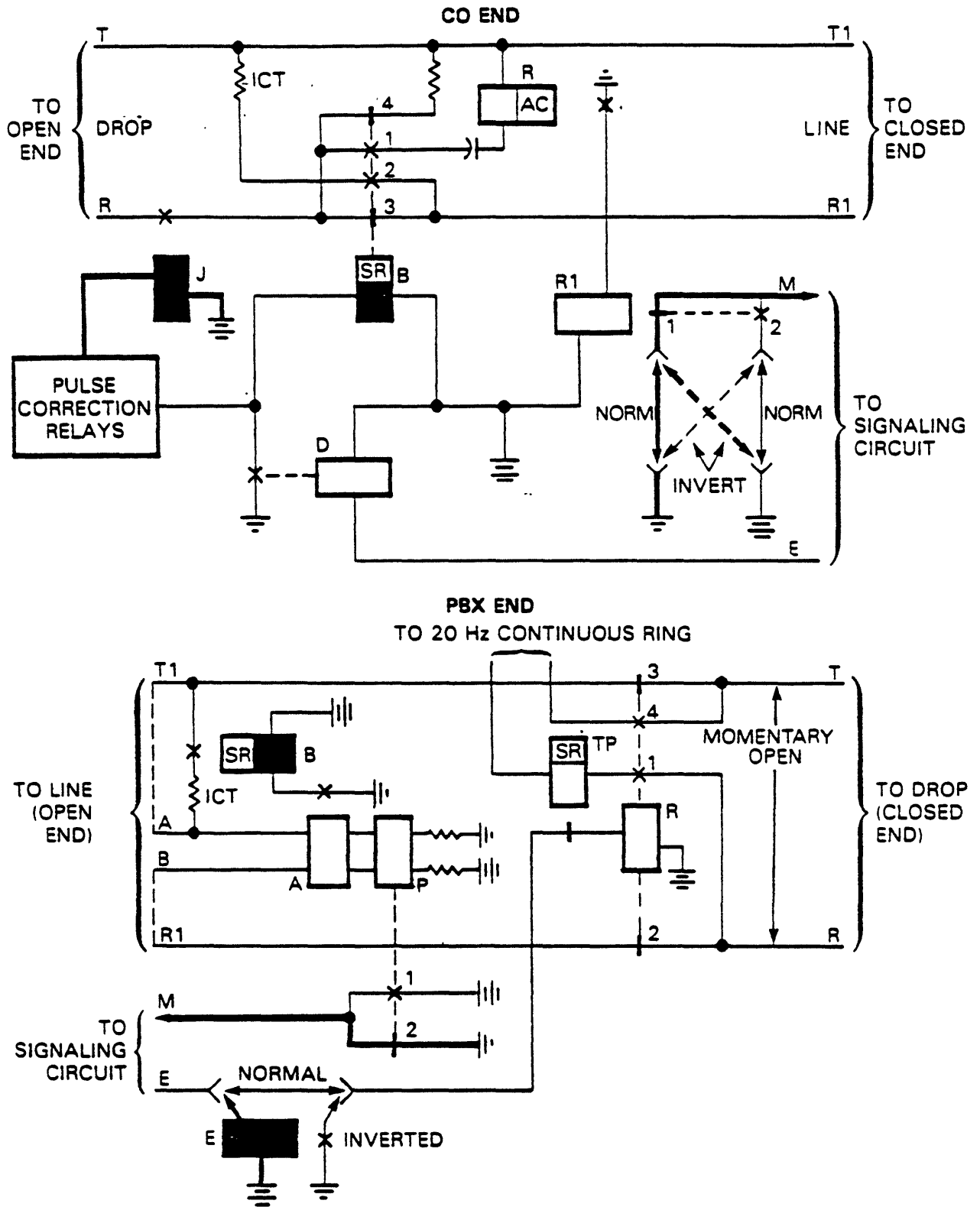




10813A083

Figure A-5b. PBX Seizure and CO Response

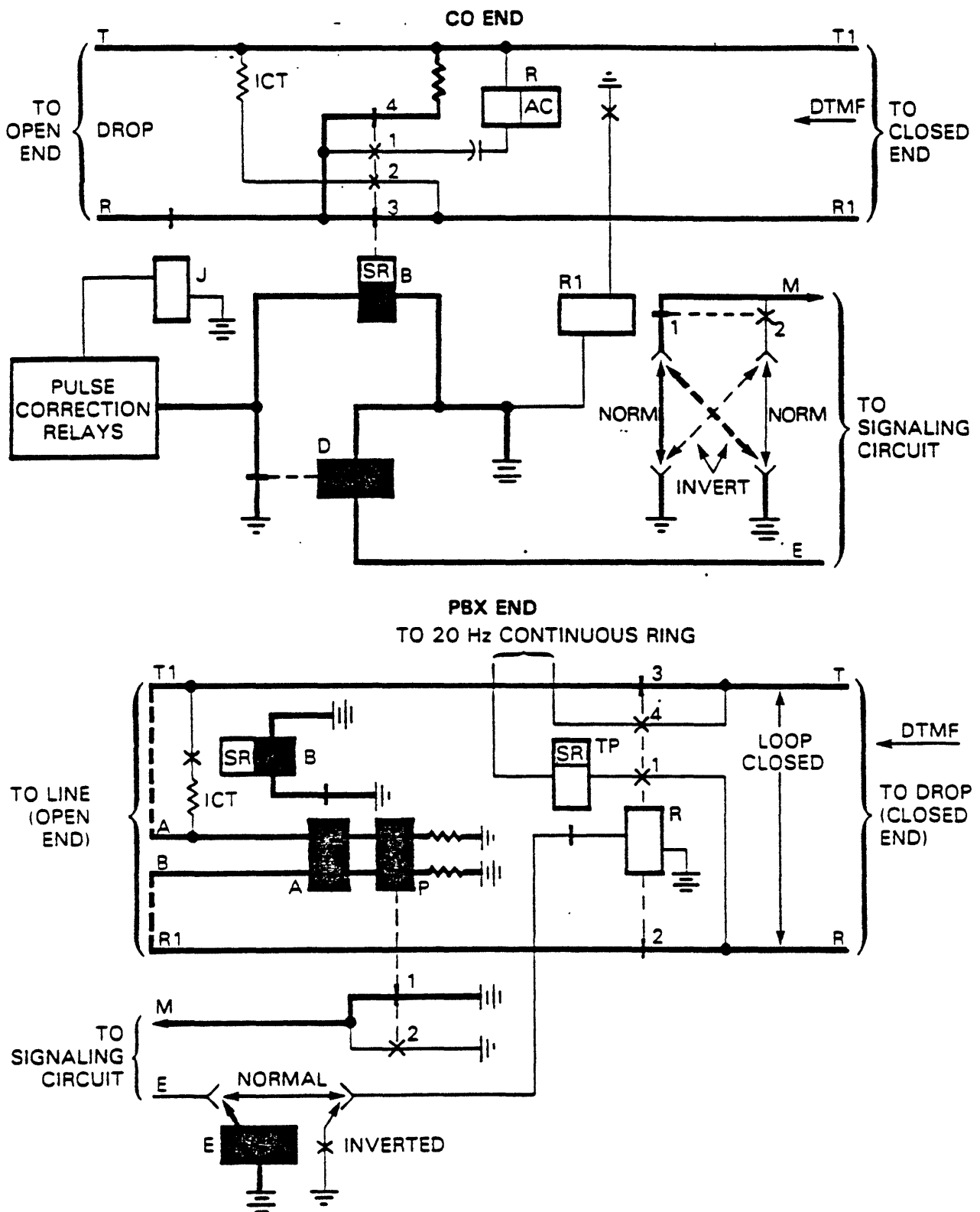
**Part I  
Trunking**



10813A084

Figure A-5c. Dial Pulse Break

**DTMF Addressing:** DTMF addressing is transmitted over the audio path and causes no change of state in the signaling leads (Figure A-5d).



10813A085

Figure A-5d. DTMF Addressing

## Part I Trunking

**Occupied Circuit:** When the distant end answers, an audio path is established and conversation can commence. The occupied condition is shown in Figure A-5e.

Note that the M lead from the CO and the E lead to the PBX do not change state during PBX call origination.

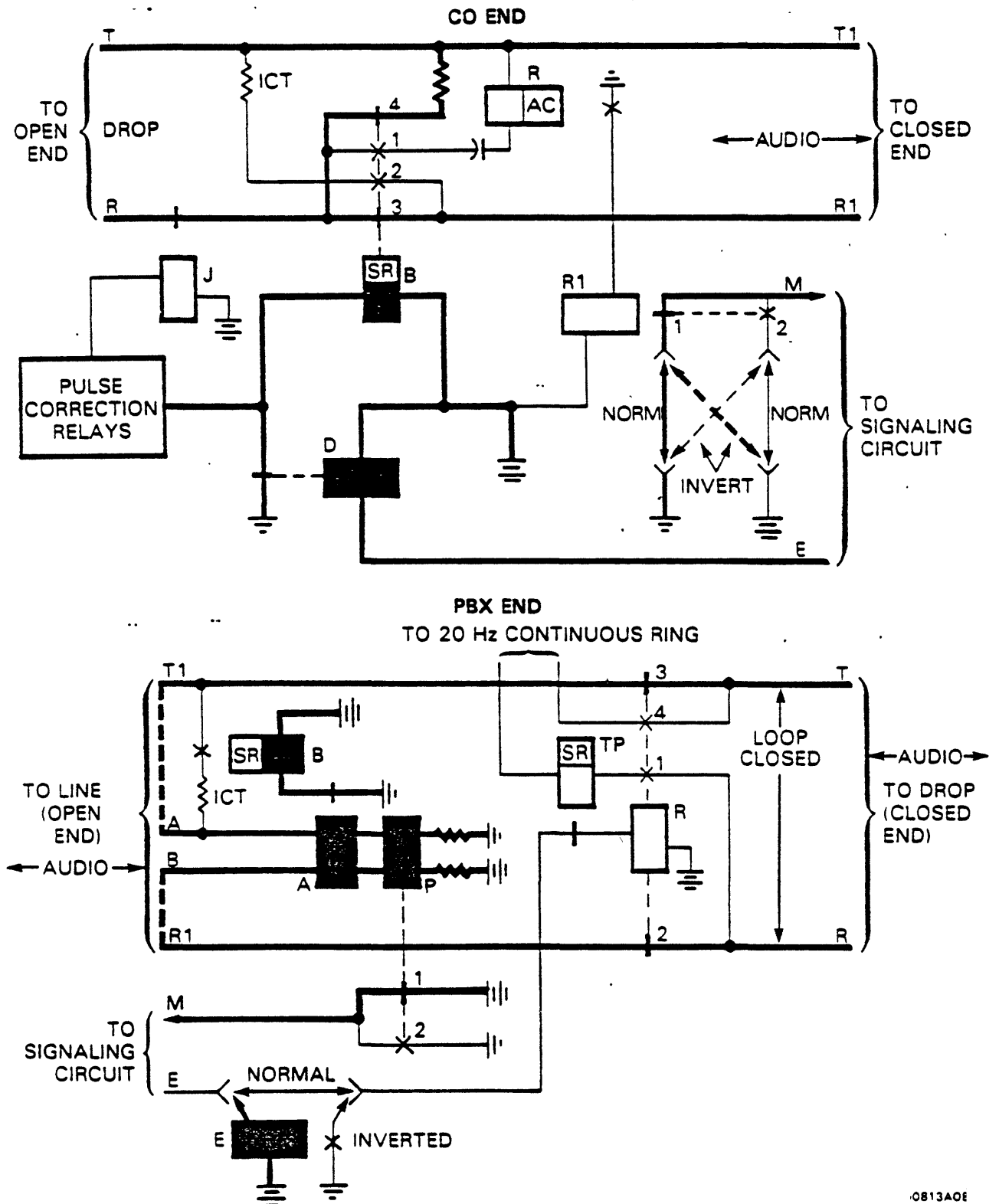


Figure A-5e. Occupied Circuit

0813A0E

**INCOMING CALL (CO ORIGINATION):** An incoming call is illustrated with reference to the following events and their associated figures:

<u>EVENT</u>	<u>FIGURE</u>
Incoming Ring	A-6a
Ring Tripping During Ring (Step 1)	A-6b
Ring Tripping During Ring (Step 2) and Ring Tripping During Rest	A-6c

**Incoming Ring:** An incoming call is initiated when the CO places 20 Hz ringing voltage, superimposed on -48 Vdc, on the line. The R relay, sensitive to ac, operates, closing the R contacts. The R1 relay follows, changing the state of the M lead and signaling the ringing toward the PBX. Ringing is applied for approximately 2 seconds, followed by a rest period of approximately 4 seconds (Figure A-6a).

The signaling circuit changes the state of the E lead toward the PBX to signal ringing. Note that the potential at point A changes from open to ground for both normal and inverted operation.

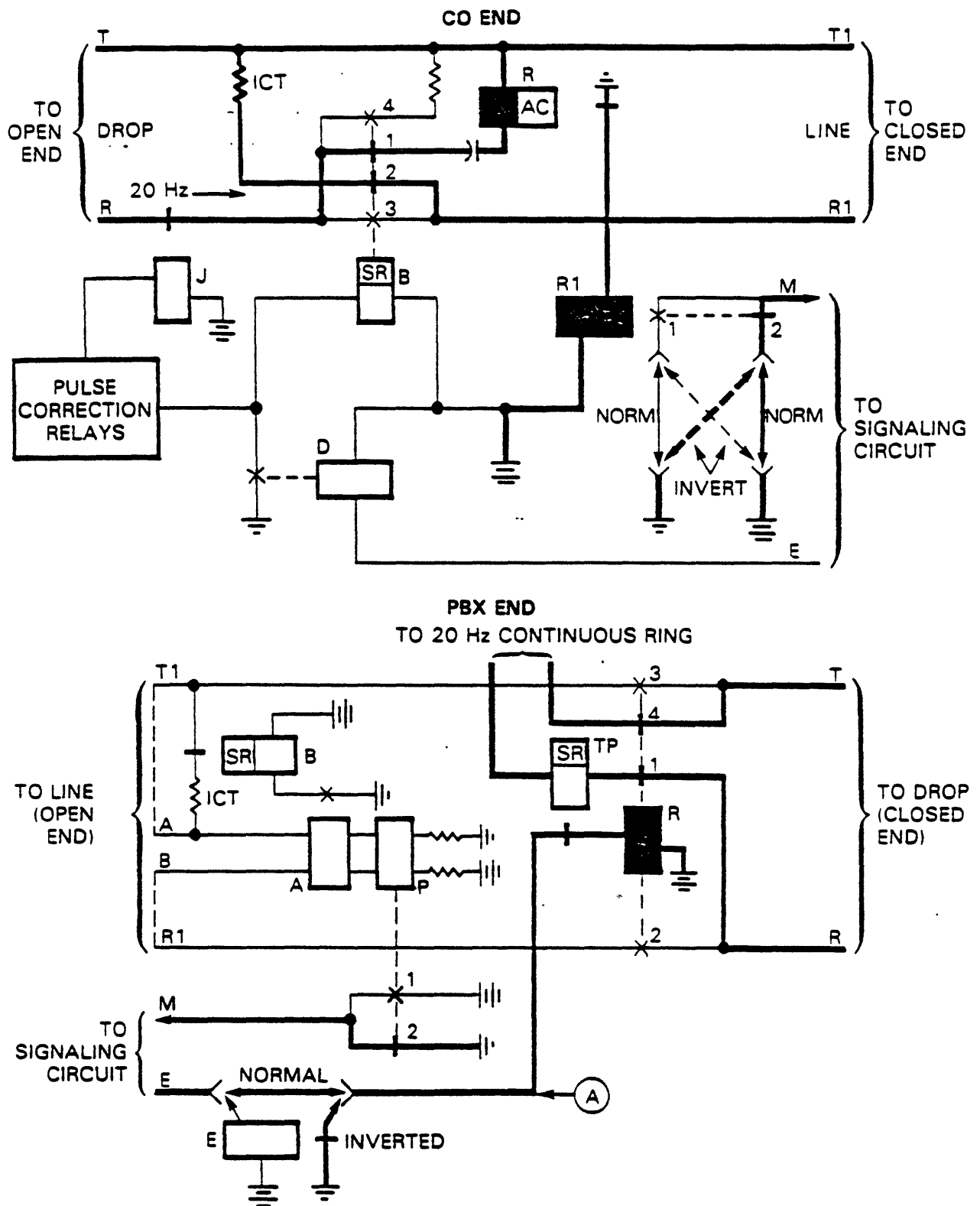
The R relay operates and closes the R(1) and R(4) contacts; this causes ringing voltage from the continuous ringing supply to be applied toward the drop side. The R(2) and R(3) contacts open, preventing ringing toward the line.

The TP relay does not operate because it is not sensitive to ac.

**Ring Tripping:** When the call is answered, the ringing voltage must be removed immediately to avoid an annoying ring in the subscriber's ear. Figure A-6b illustrates the first step of ring tripping in the case where the call is answered during the ringing period.

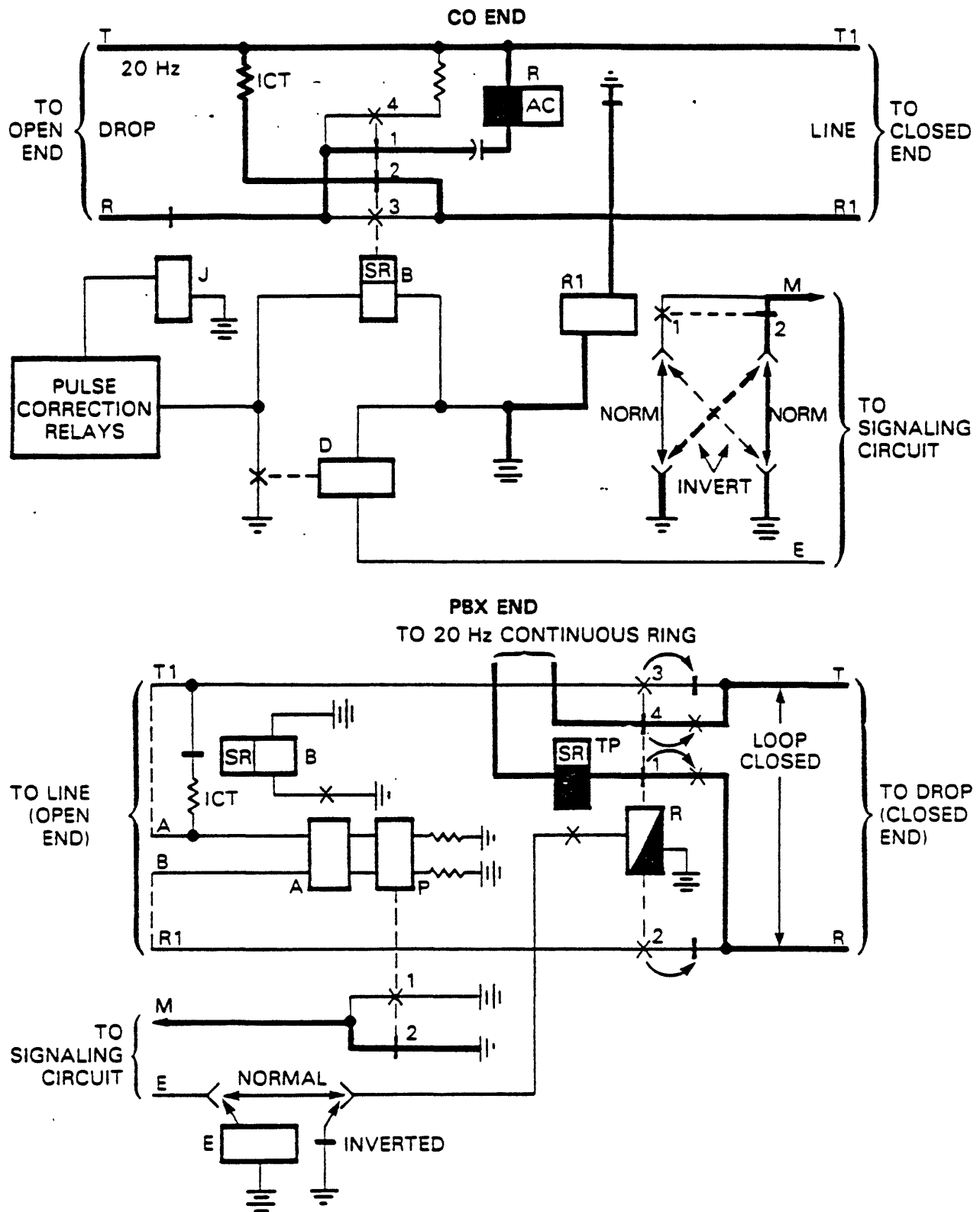
Figure A-6c illustrates ring tripping in the case where the call is answered during the rest period.

**Part I  
Trunking**



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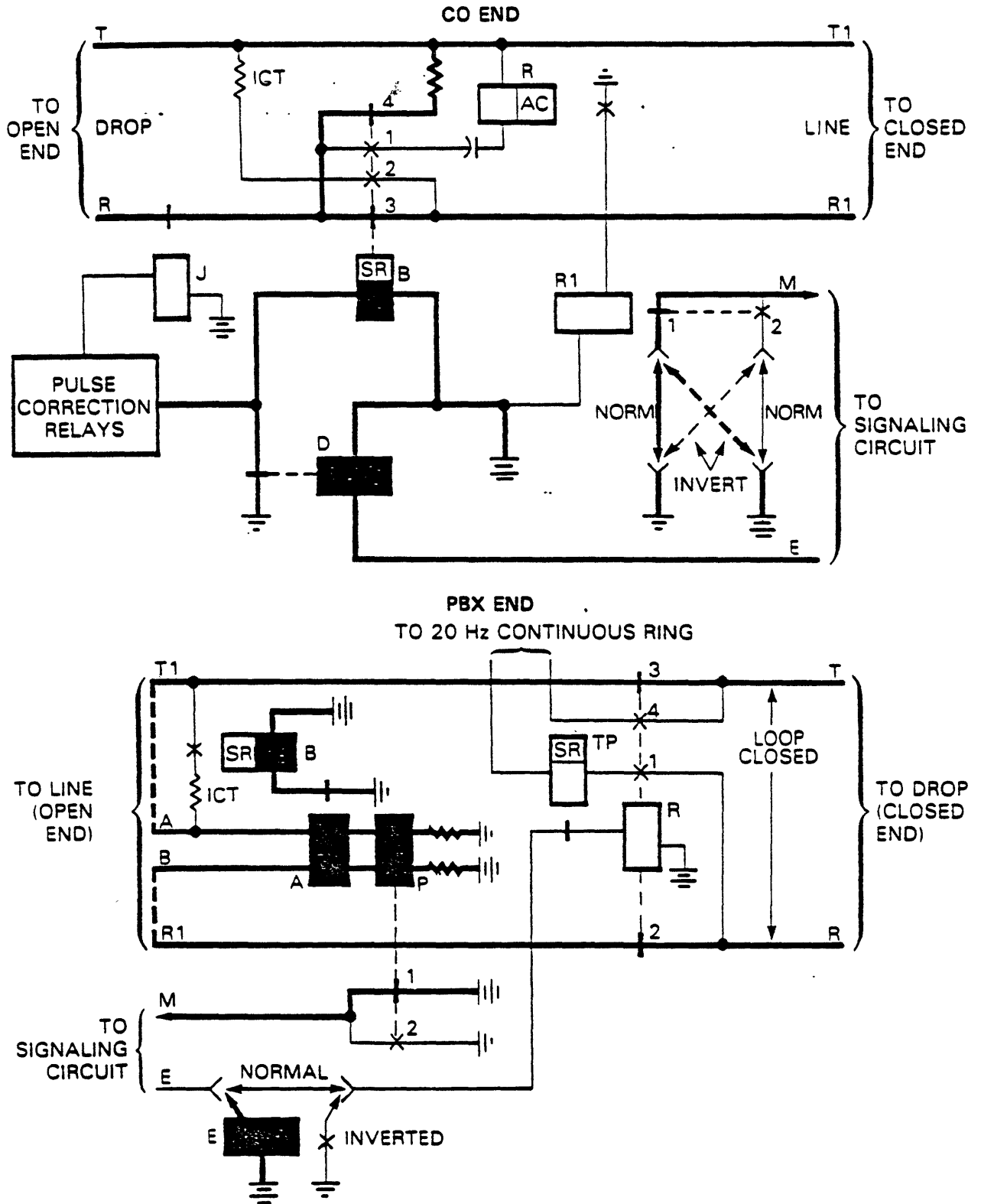
Figure A-6a. Incoming Ring



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Figure A-6b. Ring Tripping During Ring (Step 1)

**Part I  
Trunking**



10813A089

Figure A-6c. Ring Tripping During Ring (Step 2) and Ring Tripping During Rest











## APPENDIX B OVERALL GROUND START OPERATION

### OPERATION OF GROUND START SIGNALING

Overall operation of ground start signaling will be discussed with reference to the simplified circuit schematics in Figure B-1, Figures B-2a through B-2e, Figures B-3a through B-3d, and Figure B-4.

The symbols and conventions used in the schematics are the same as those used to illustrate loop start operation. These symbols and conventions are illustrated in Table B-1.

**Table B-1. Symbols and Conventions**

SYMBOL	EXPLANATION
	Heavy line. Indicates active portion(s) of the circuit for the operation being described.
	Light line. Indicates inactive circuitry for the operation being described.
	Light relay symbol. No current in relay windings, relay released.
	Dark relay symbol. Current in relay windings, relay operated.
	Relay in transition.
	Relay contact closed.
	Relay contact open.
	Audio path.

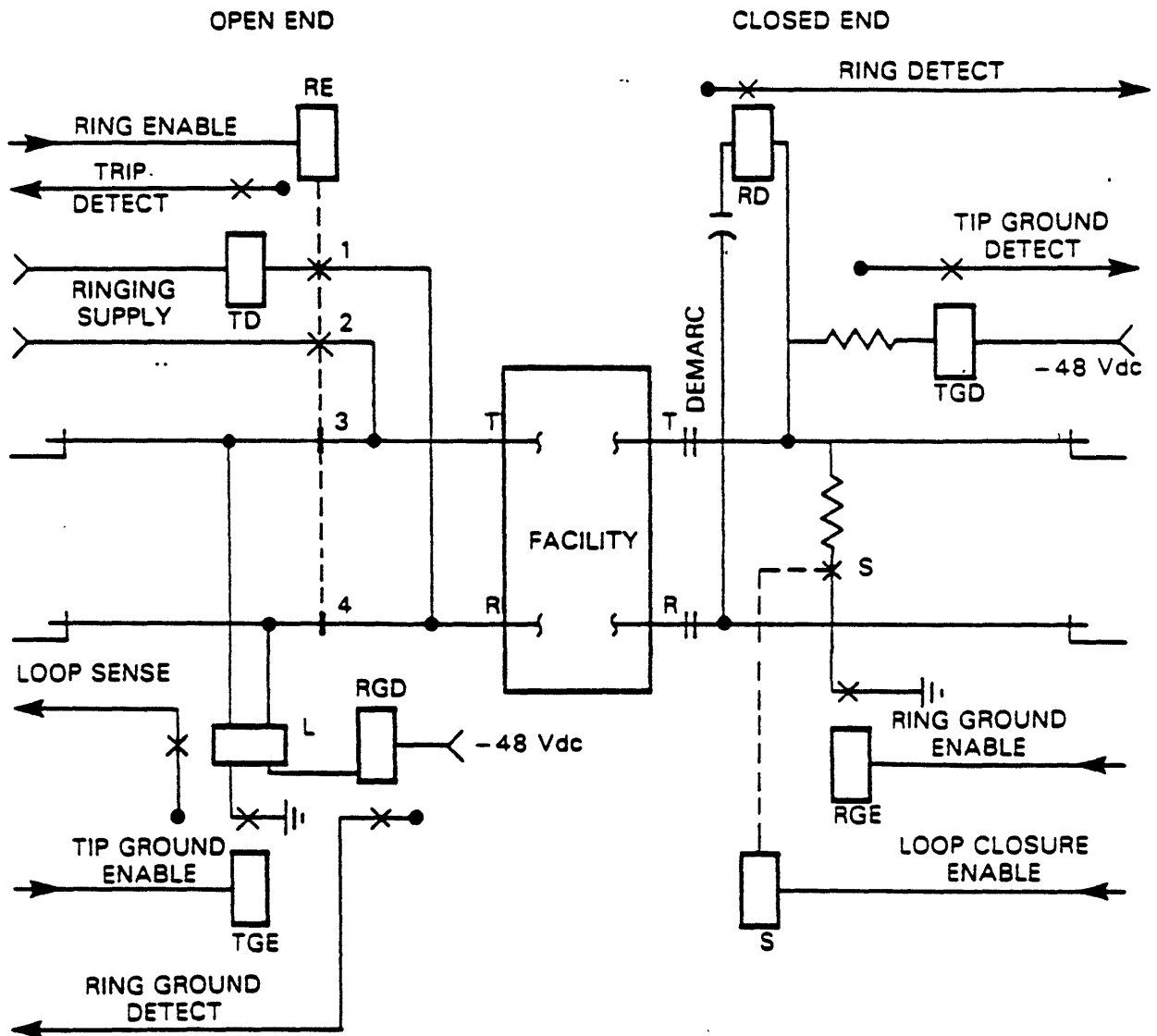
## Part I Trunking

**IDLE CONDITION:** The idle condition is represented in Figure B-1. All relays are in the released state and the relay contacts are as shown.

The circuitry used for ground start operation is very similar to that used in loop start. Ground start uses four additional leads:

- Open (CO) End
  - Ring Ground Detect (RGD)
  - Tip Ground Enable (TGE)
- Closed (PBX) End
  - Ring Ground Enable (RGE)
  - Tip Ground Detect (TGD)

These four additional leads provide for more positive supervision and resolve the glare, start dial, and distant-end disconnect problems encountered in loop start signaling.



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Figure B-1. Idle Condition

**OUTGOING CALL (PBX ORIENTATION):** An outgoing call is described with reference to Figures B-2a through B-2e.

EVENT	FIGURE
PBX Seizure	B-2a
CO Response	B-2b
PBX Loop Closure and DTMF Addressing	B-2c
Dial Pulsing	B-2d
Occupied Circuit	B-2e

**PBX Seizure:** The PBX seizes the circuit by grounding the ring lead toward the CO, as shown in Figure B-2a. To do this, the PBX operates the relay via the ring ground enable lead. The RGE contacts close, causing current to flow over the ring lead between the PBX and the CO. Current over the ring lead operates the RGD relay, informing the CO of the seizure via the ring ground detect lead. This prevents the CO from seizing the circuit for an incoming call. The L relay remains released, as it requires current in both the tip and ring leads in order to operate.

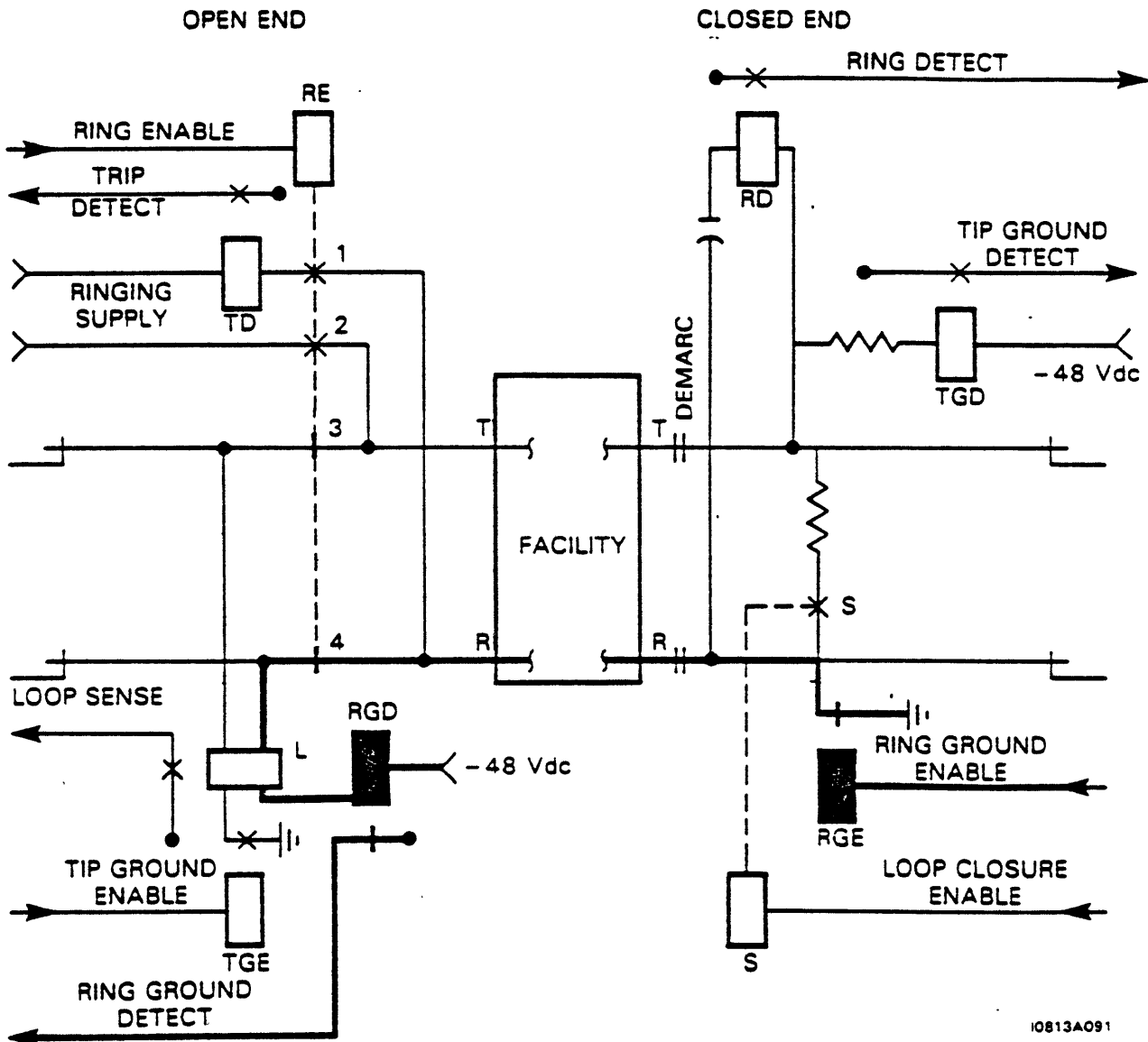


Figure B-2a. PBX Seizure

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## Part I Trunking

**CO Response:** The CO responds (Figure B-2b) to the seizure by grounding the tip lead toward the PBX, establishes an audio path toward the PBX and transmits dial tone. In order to ground the tip lead, the CO operates the TGE relay via the tip ground enable lead.

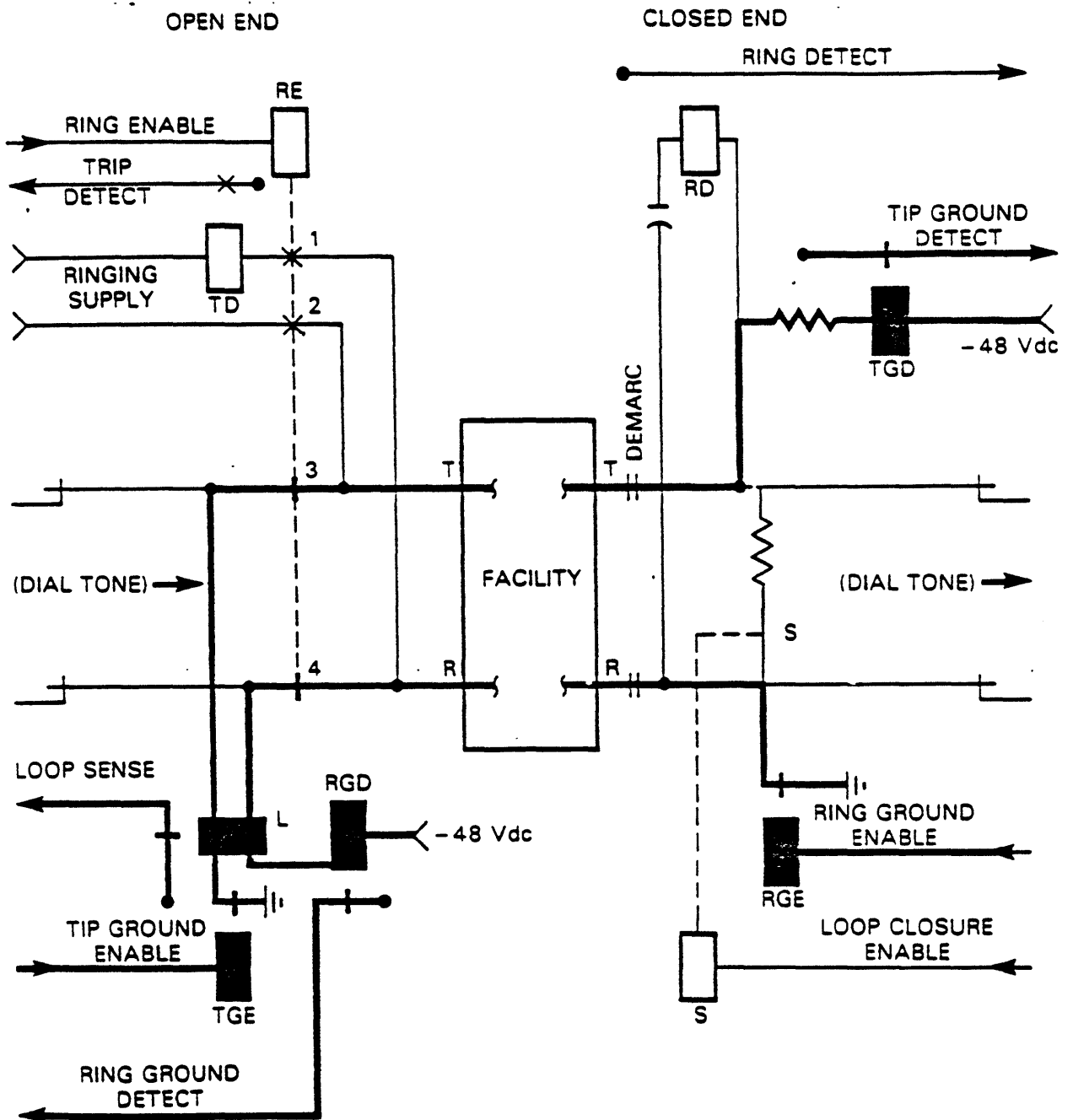


Figure B-2b. CO Response

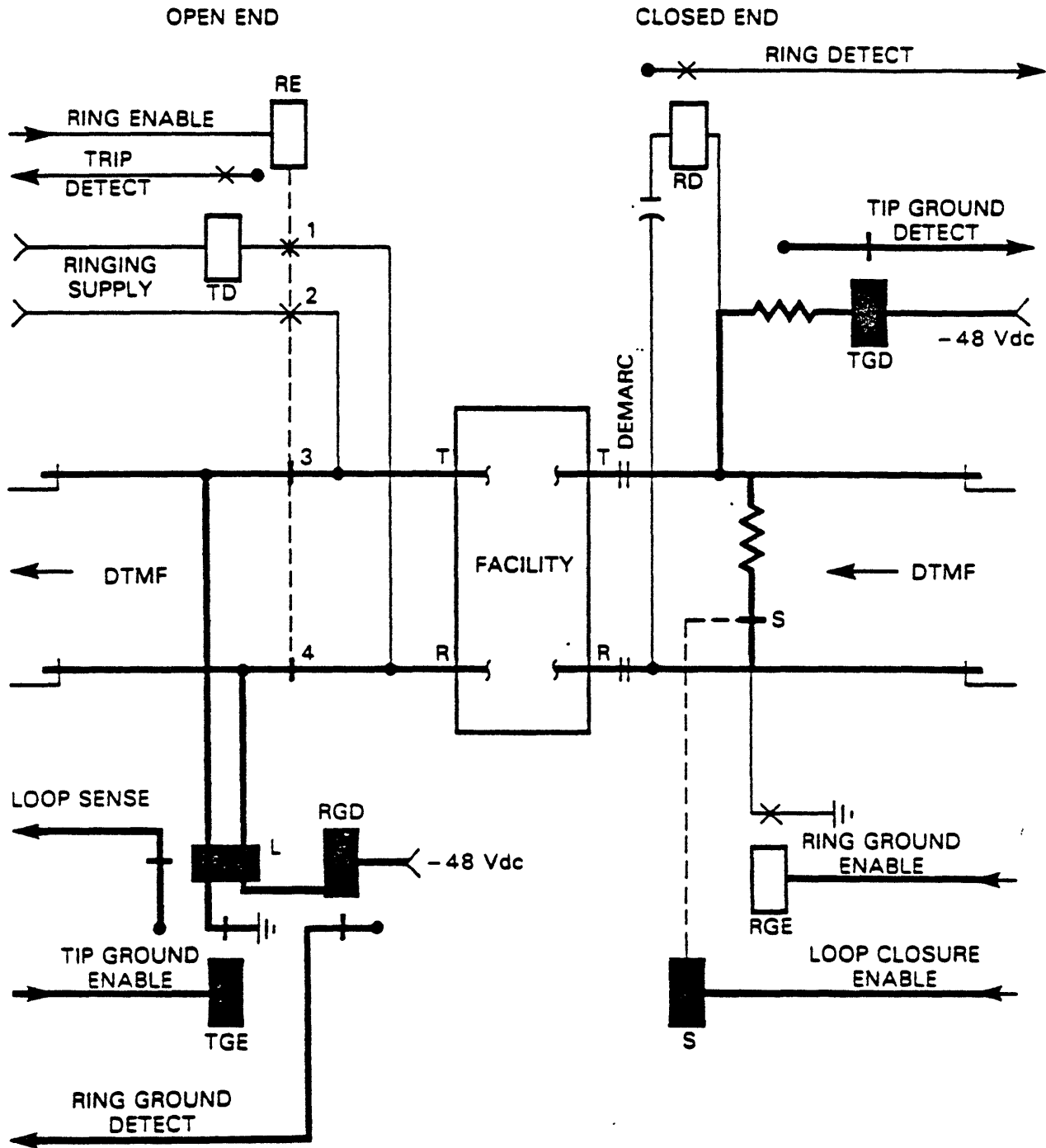
10813A092

**PBX Loop Closure and DTMF Addressing:** Upon sensing tip ground over the tip ground detect lead, the PBX removes the ring ground by releasing the RGE relay via the ring ground enable lead and simultaneously closes the loop by operating the S relay via the loop closure enable lead (Figure B-2c)

With the audio path established toward the CO, the PBX may proceed to send address information to the CO.

In sendedized operation, the PBX begins sending digits based upon detecting the ground on the tip. In non-sendedized operation, dial tone is sent to the station over the audio path, informing the subscriber that the CO is ready to receive address information.

DTMF addressing is transmitted to the CO over the audio path.

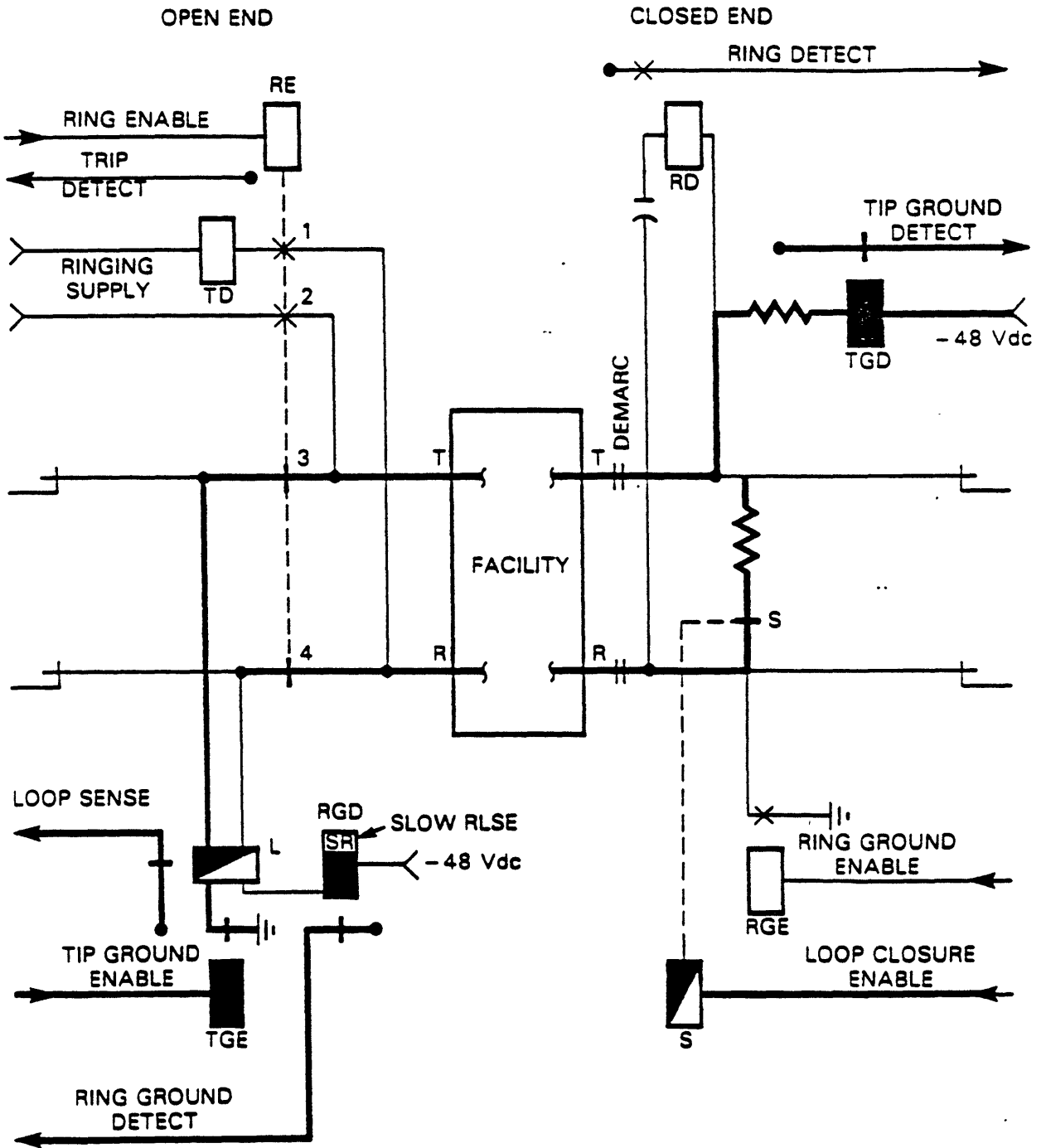


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**Figure B-2c. PBX Loop Closure and DTMF Addressing**

**Part I  
Trunking**

**Dial Pulsing:** As in loop start operation, dial pulse information is transmitted toward the CO by alternately opening and closing the loop via the loop closure enable lead and the S relay (Figure B-2d). The L relay in the CO follows the S relay in the PBX and the CO recognizes dial pulses via the loop sense lead. Dial pulse breaks are not interpreted by the CO as PBX disconnect due to their short duration.

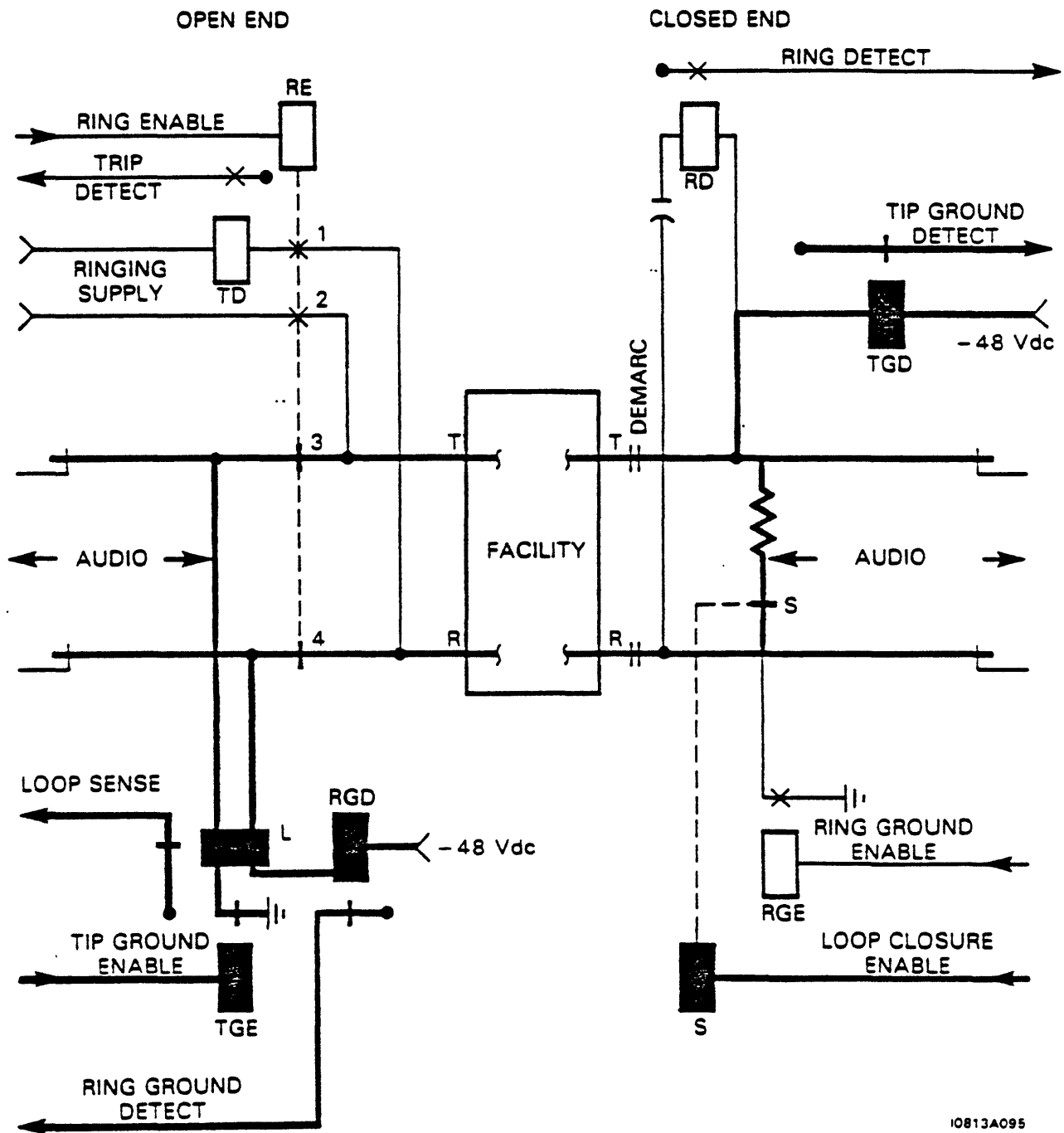


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Figure B-2d. Dial Pulsing

**Occupied Circuit:** The occupied circuit is represented in Figure B-2e. The relays are functioning as follows:

- The S relay is holding up the call in the PBX.
- The L relay is holding up the call in the CO.
- The TGD relay is waiting for the CO disconnect signal.
- The L relay is waiting for the PBX disconnect signal.



10813A095

**Figure B-2e. Occupied Circuit**

**Part I  
Trunking**

**INCOMING CALL (CO ORIGINATION):** An incoming call is described with reference to Figures B-3a through B-3d:

EVENT	FIGURE
CO Seizure	B-3a
Incoming Ring	B-3b
Ring Tripping During Ring	B-3c
Answer During Rest	B-3d

**CO Seizure:** The CO seizes the line by operating the TGE relay via the tip ground enable lead. Current in the tip lead operates the TGD relay, and the PBX recognizes the incoming seizure via the tip ground detect lead.

The PBX will not seize the circuit for an outgoing call as long as the TGD relay is operated.

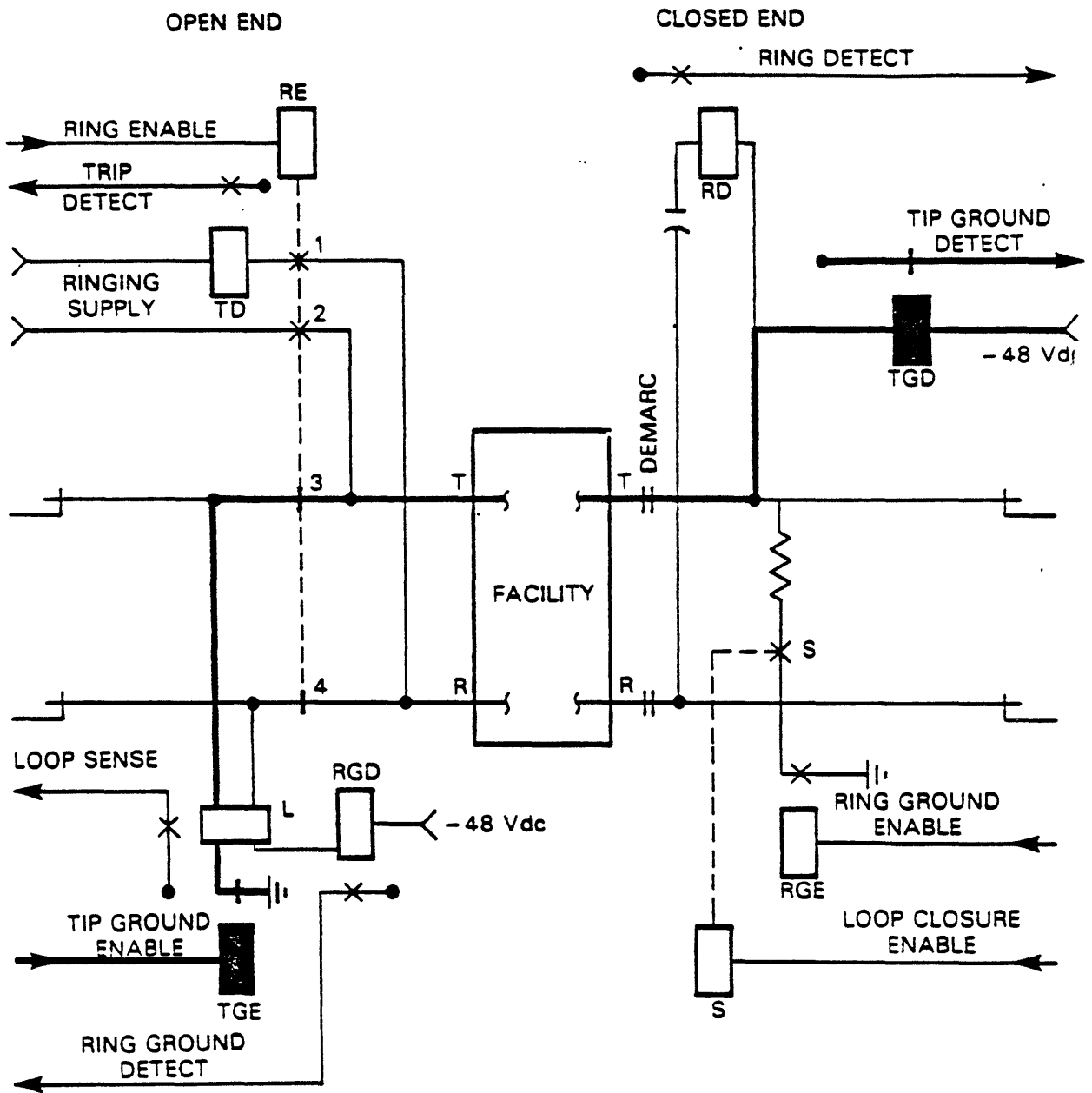


Figure B-3a. CO Seizure

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## Part I Trunking

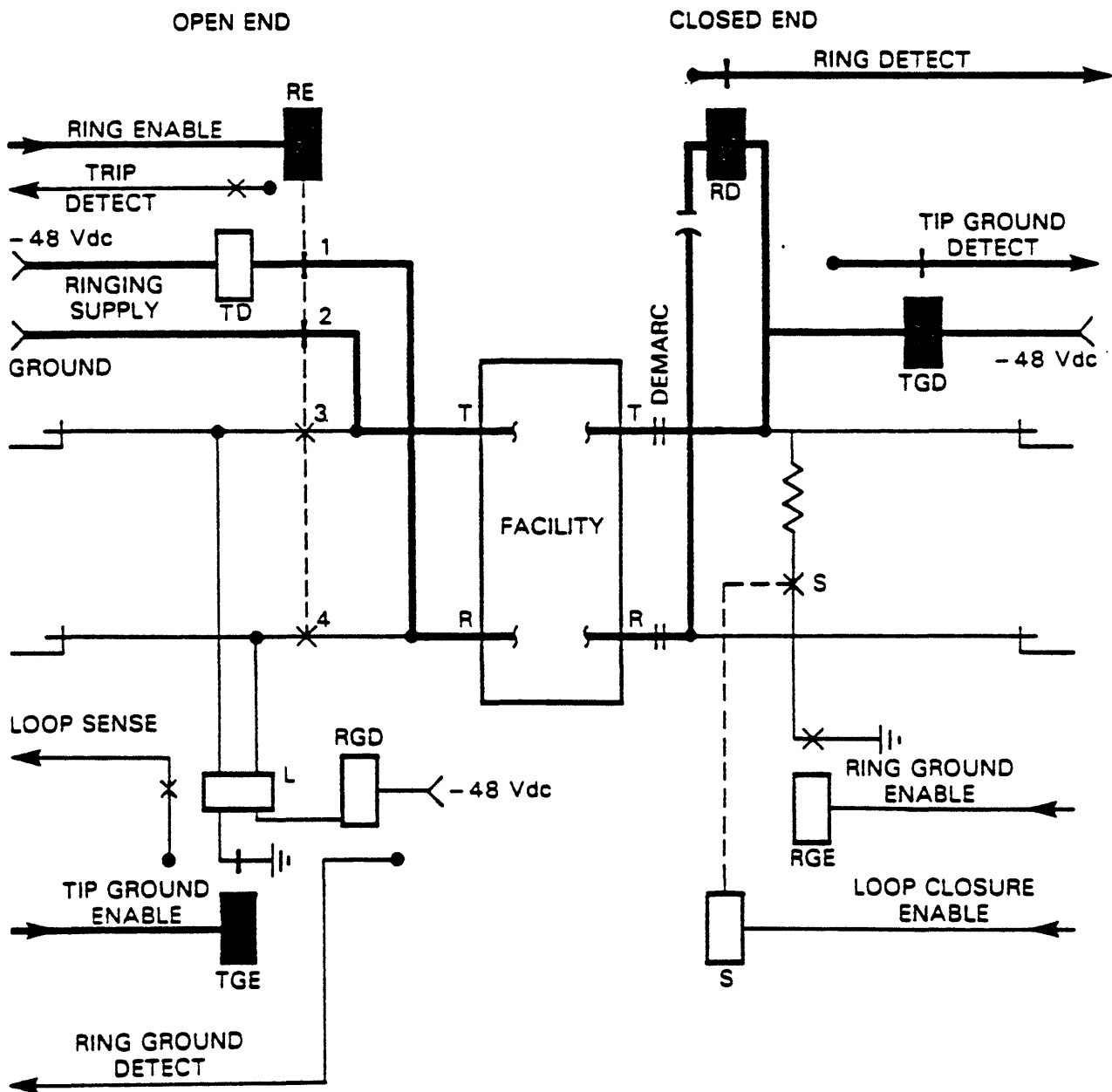
**Incoming Ring:** The CO initiates ringing by operating the RE relay via the ring enable lead (Figure B-3b). The RE(1) and RE(2) contacts close, connecting the continuous ringing supply to the circuit. A 20 Hz ringing voltage, superimposed on  $-48$  Vdc, is applied to the ring lead. The grounded tip lead provides a return path for the 20 Hz ac.

Contacts RE(3) and RE(4) are open to prevent ringing toward the call originator.

In the PBX, ringing voltages operate the RD relay, and the PBX detects ringing via the ring detect lead.

The TGD relay remains operated by current flowing over the tip lead from its  $-48$  Vdc supply to the ground in the ringing supply. This is important, as it prevents the PBX from seizing the circuit during the rest period, which would result in a glare condition.

Ringing continues (2 seconds ring, 4 seconds rest) until the PBX answers or until the distant end disconnects.



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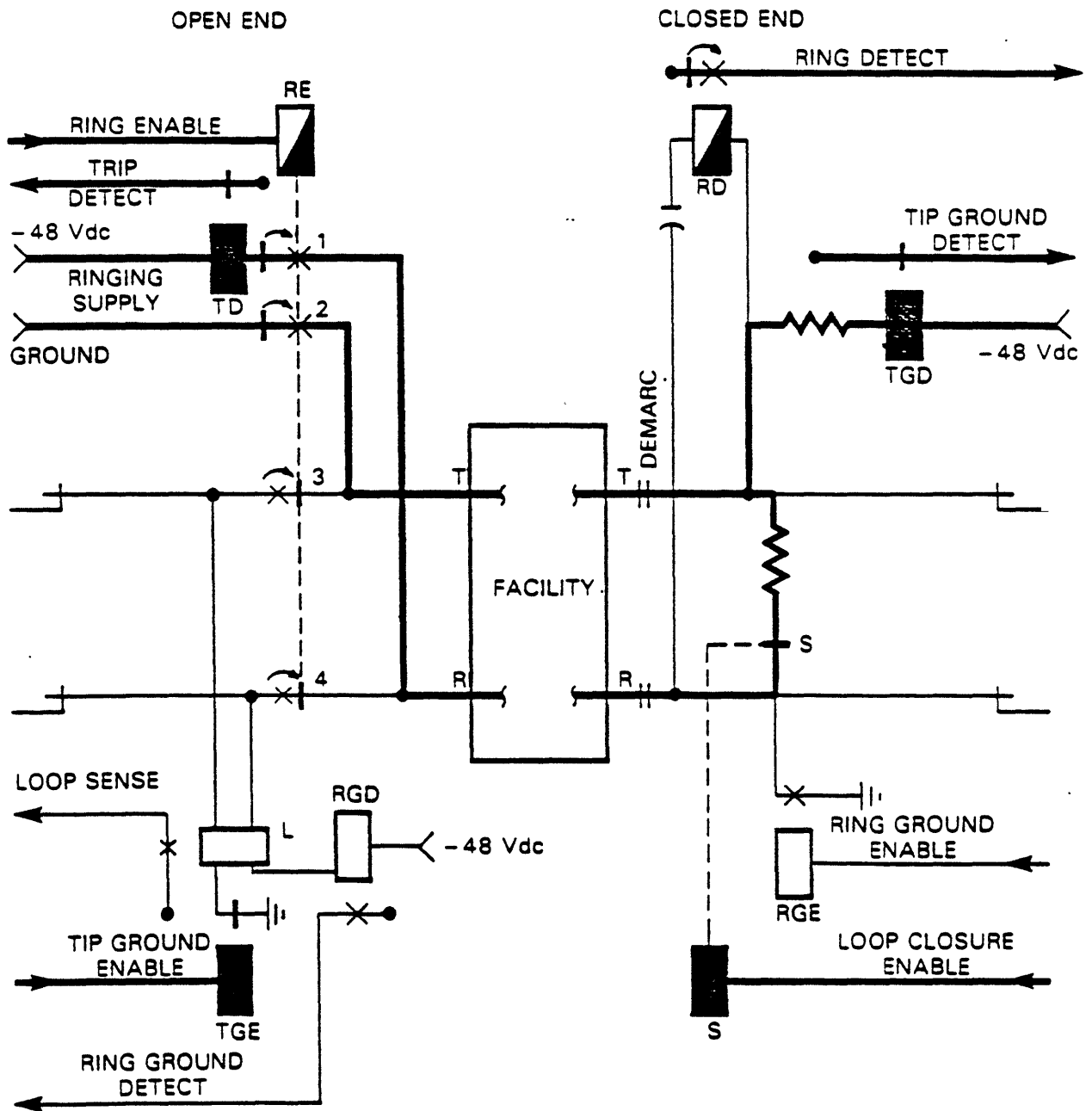
Figure B-3b. Incoming Ring

## Part I Trunking

**PBX Answer:** The PBX answers by operating the S relay via the loop closure enable lead, thus closing the loop toward the CO. The PBX may answer during either the ringing period or the rest period.

**Ring Tripping During Ring:** If the PBX answers during the ringing period, it is important that the ring be tripped immediately in order to avoid 20 Hz ringing in the called party's ear

The PBX answers by operating the S relay via the loop closure enable lead, closing the loop toward the CO (Figure B-3c). Current flowing over the tip lead operates the TD relay. The CO recognizes the PBX answer via the trip detect lead and releases the RE relay via the ring enable lead. Contacts RE(1) and RE(2) open, removing the ringing supply. Contacts RE(3) and RE(4) close, completing the loop through the L relay in the CO.



10813A098

Figure B-3c. Ring Tripping During Ring

**Answer During Rest:** The case where the PBX answers during the rest period is represented in Figure B-3d. (Figure B-3d also shows circuit operation after ring tripping during ring.)

The TGE lead has remained operated since incoming seizure. Loop current causes the L relay to operate and the CO extends the audio path to the PBX. The circuit is now in the occupied condition.

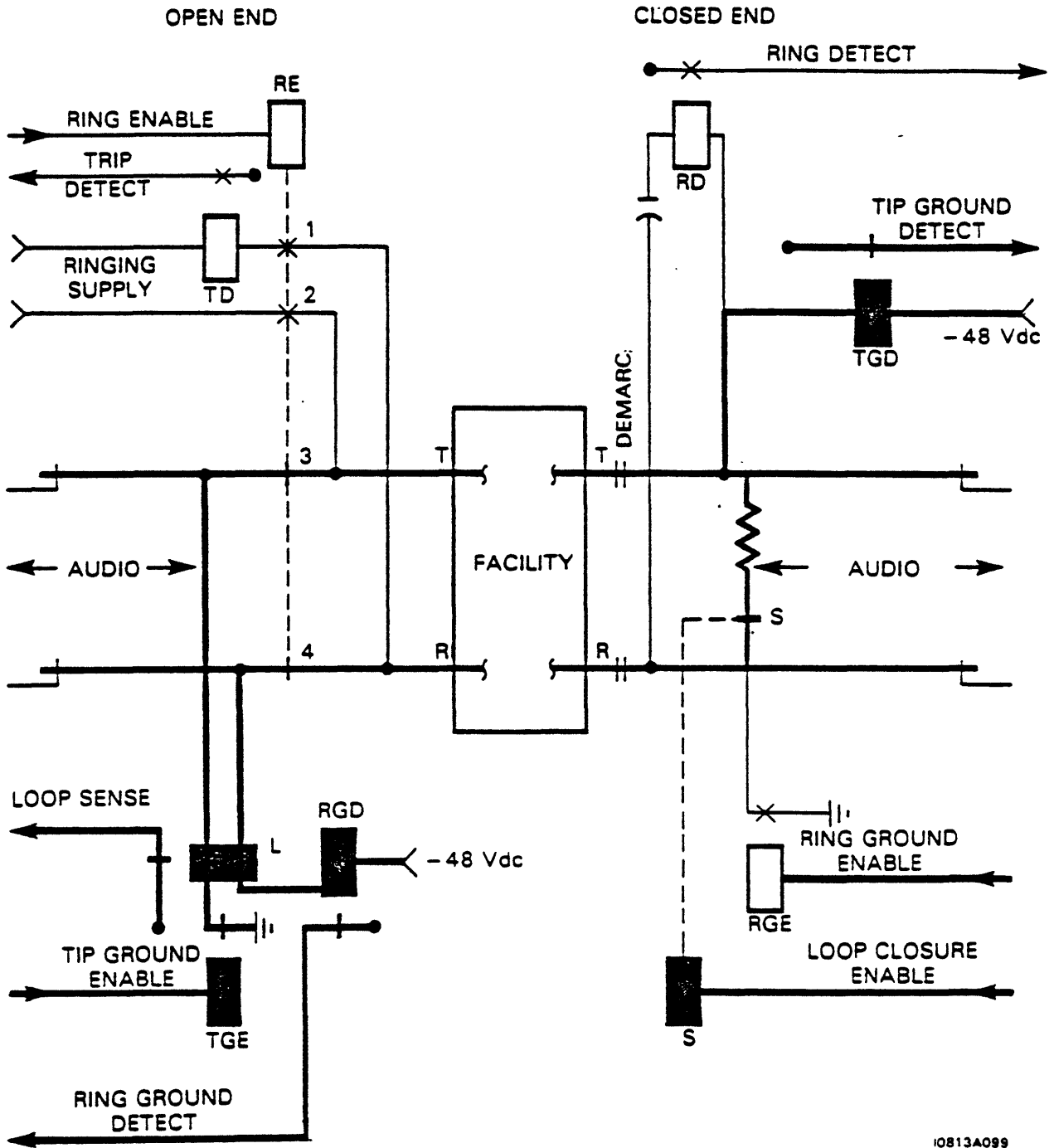


Figure B-3d. Answer During Rest

**Part I  
Trunking**

**CALL DISCONNECT:** Disconnect is initiated when either end goes on-hook. Both near-end disconnect and distant-end disconnect are shown in Figure B-4

**Near-end (PBX) Disconnect:** Near-end disconnect is initiated when the PBX releases the S relay via the loop closure enable lead. The S contacts go from closed to open, and current ceases flowing in the loop.

Lack of current in the loop releases the RGD and L relays in the CO. Upon recognizing the disconnect, the CO releases the TGE relay via the tip ground enable lead, and the circuit is returned to the idle condition.

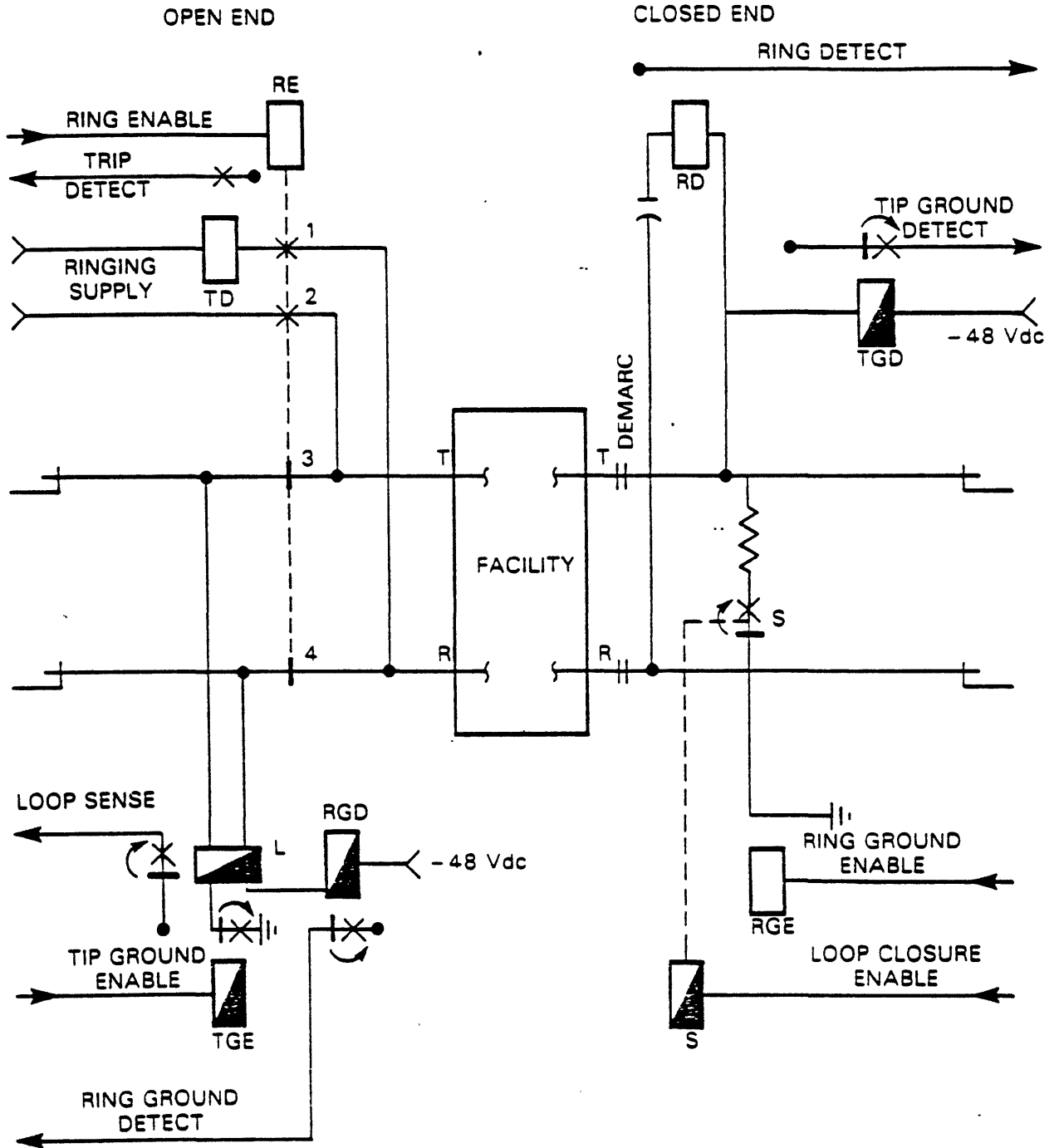


Figure B-4. Disconnect

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**Distant-end (CO) Disconnect:** Upon detecting disconnect at the distant end, the CO releases the TGE relay via the tip ground enable lead. Current ceases in the loop, and the L and RGD relays are released.

In the PBX, the TGD relay releases, informing the PBX of the distant-end disconnect via the tip ground detect lead.

Note that ground start operation provides positive signaling to the PBX when the distant end goes on-hook via the tip ground enable lead in the CO and the tip ground detect lead in the PBX.

## **OPERATION OF GROUND START TO E&M INTERFACE**

Overall operation of the ground start to E&M interface is discussed with reference to simplified circuit schematics contained in Figures B-5a through B-5f (outgoing call) and Figures B-6a through B-6d (incoming call).

**OUTGOING CALL (PBX ORIENTATION):** An outgoing call is illustrated by breaking the call into the following events and their associated figures:

<b>EVENT</b>	<b>FIGURE</b>
Idle Condition	B-5a
PBX Seizure	B-5b
CO Response	B-5c
PBX Loop Closure and DTMF Addressing	B-5d
Dial Pulse Break	B-5e
Occupied Circuit	B-5f

**Idle Condition:** The idle condition is represented in Figure B-5a.

**PBX Seizure:** The PBX seizes the circuit by placing a ground on the ring lead (Figure B-5b). The RGD relay operates, and the M relay follows. The M(1) contacts open and the M(2) contacts close, placing battery on the M lead.

At the CO end, the E lead goes from open to ground in response to battery on the M lead at the PBX. The E relay operates, closing the E(2) contacts, grounding the ring lead. The ring ground seizure at the PBX is thus duplicated at the CO end.

**CO Response:** The CO responds to the ring ground by grounding the tip lead (Figure B-5c). The TGD relay operates, simultaneously removing ring ground toward the CO (TGD(2) opens) and closing the loop toward the CO (TGD(1) closes). The CO will attach a dial tone generator upon recognizing loop closure.

The TGD(3) contacts close, causing the M relay to operate, placing battery on the M lead toward the facility. The E lead at the closed end goes from open to ground.

In Figure B-5c, the E relay is shown in the transition from released to operated.

**PBX Loop Closure and DTMF Addressing:** In Figure B-5d, operation of the E relay is shown to be complete. The E contacts close, causing current to flow in the wink detect circuit (wink-on-ring operation) or the E lead re-seizure detect circuit (ring-after-ring operation). As will be discussed later, the steady seizure of the E lead does not activate either of these circuits.

The TGE relay operates, closing the TGE contacts, sending a tip ground toward the PBX and completing the loop. The A relay operates, closing the A(1) and A(2) contacts. An audio path is now established and DTMF address signaling may proceed.

**Part I  
Trunking**

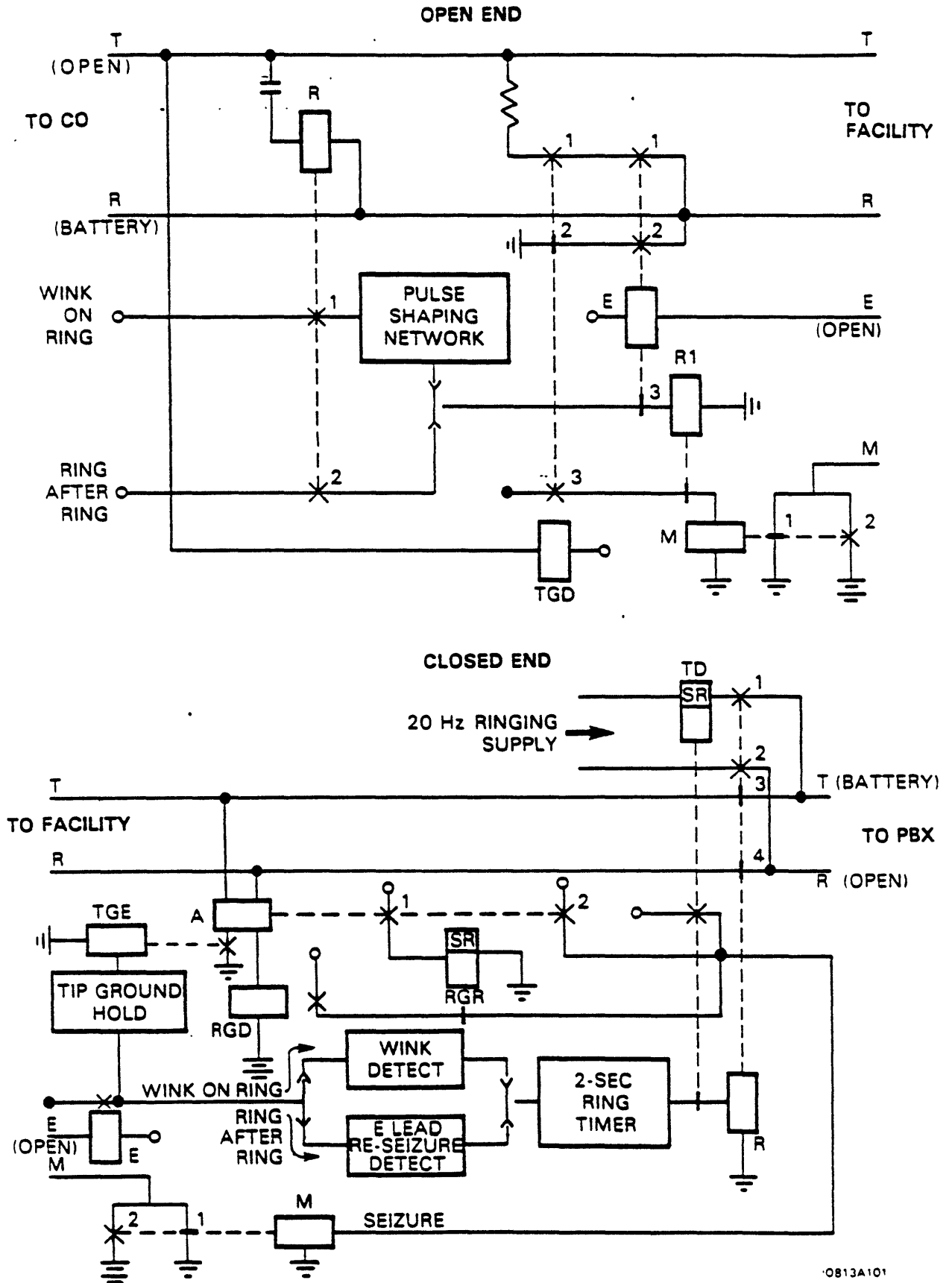
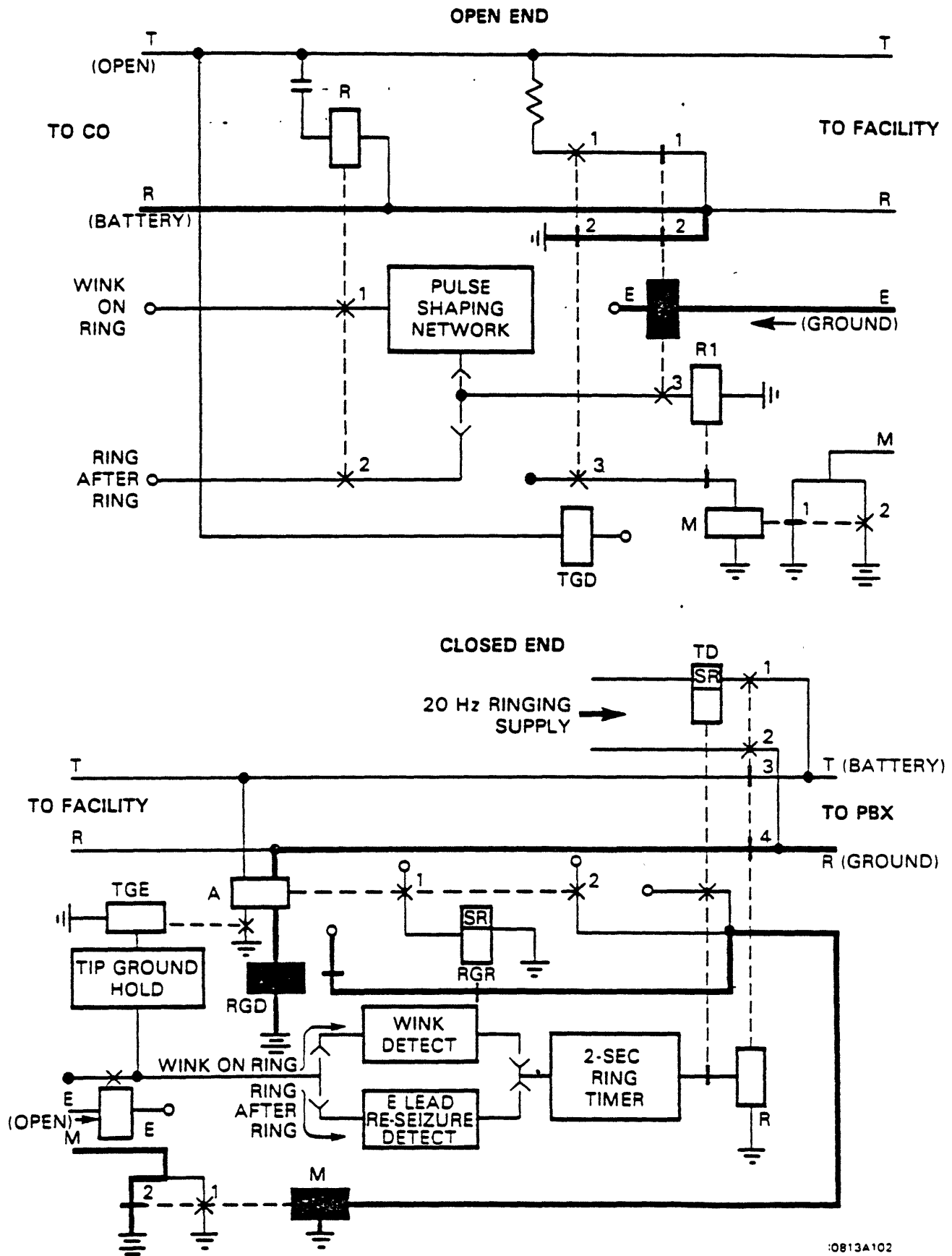


Figure B-5a. Idle Condition



10813A102

**Figure B-5b. PBX Seizure**

**Part I  
Trunking**

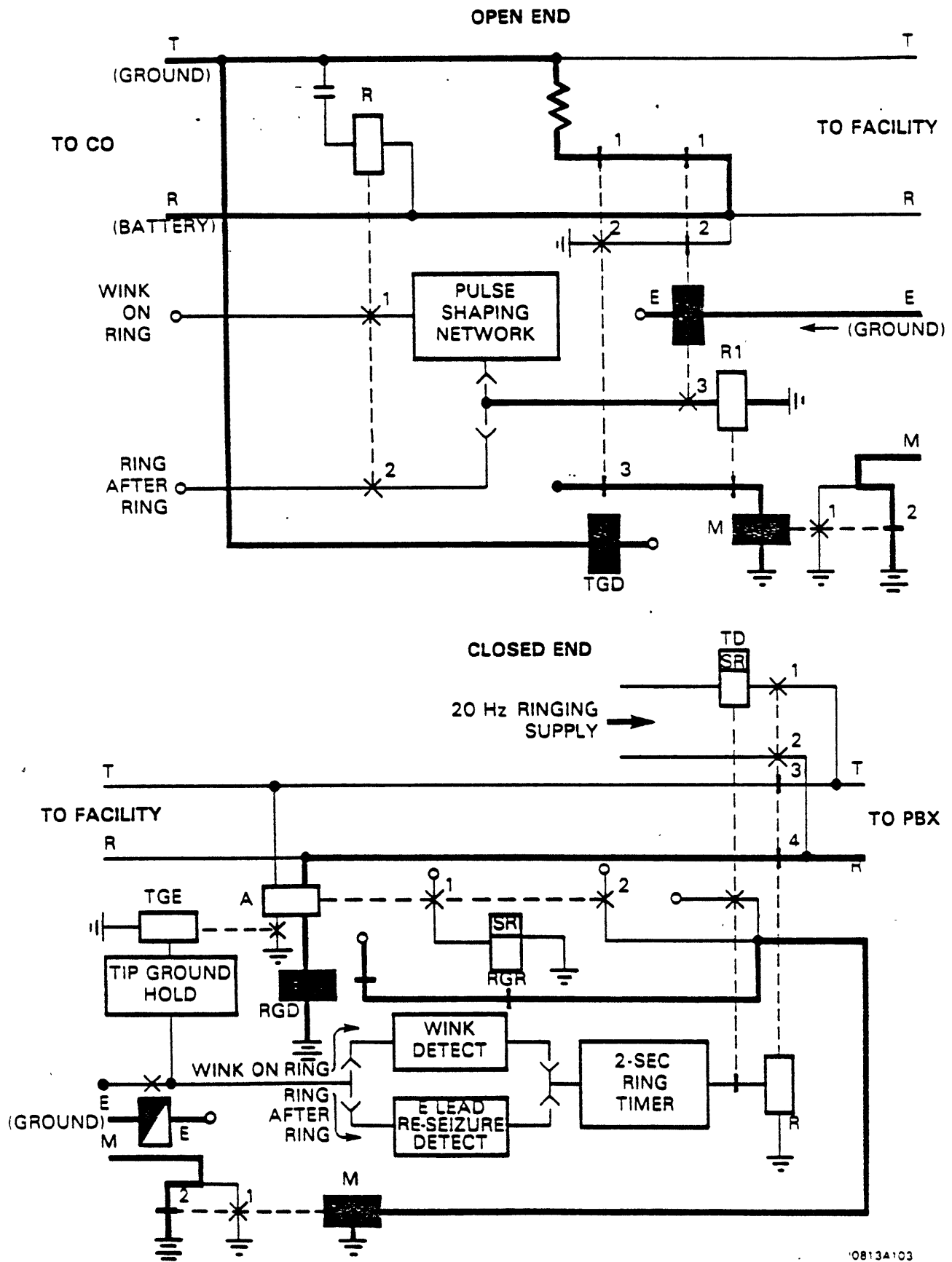
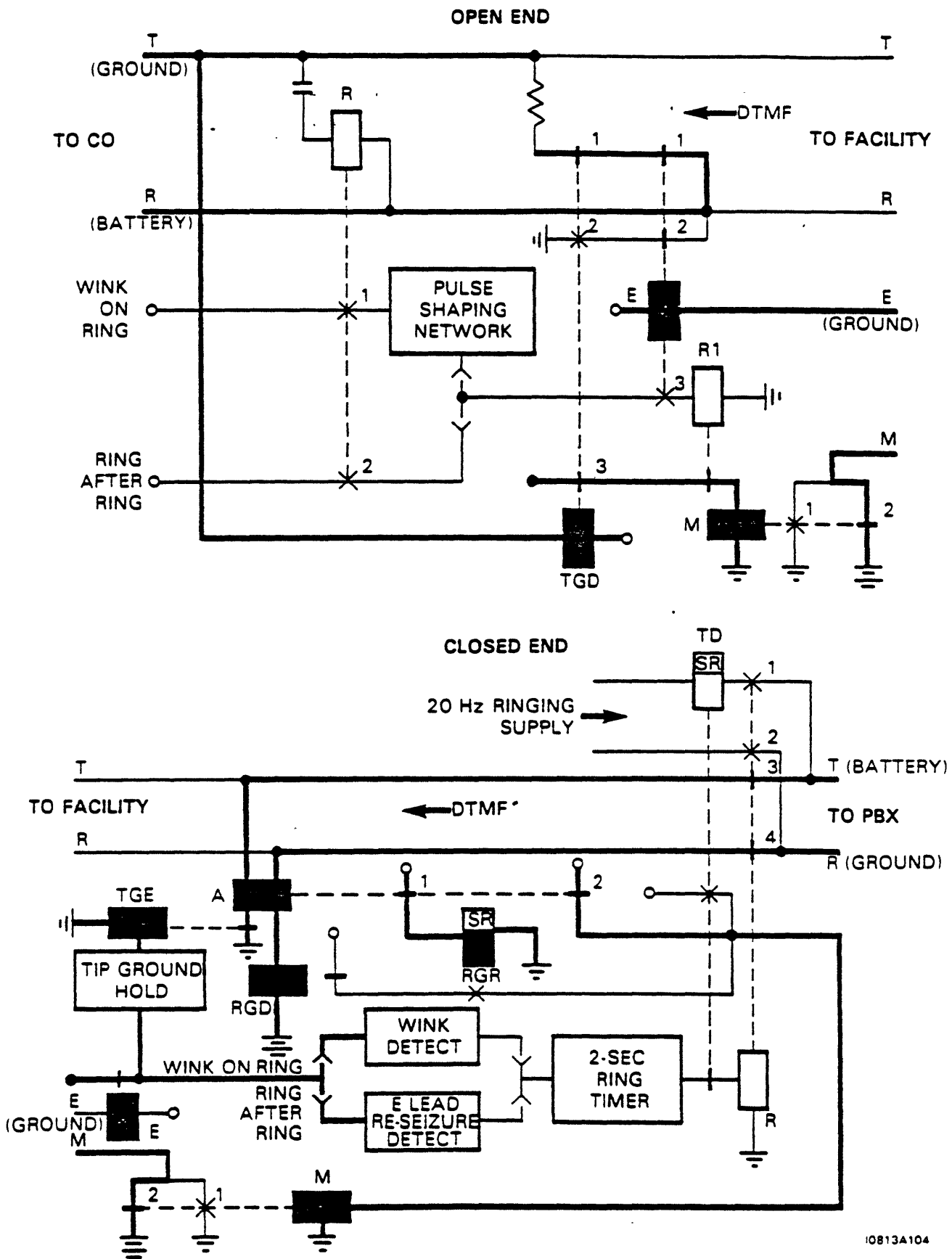


Figure B-5c. CO Response





10813A104

Figure B-5d. PBX Loop Closure and DTMF Addressing

## Part I Trunking

**Dial Pulse Break:** The PBX opens and closes the loop toward the signaling equipment in dial pulse addressing (Figure B-5e). The A relay releases in response to opening of the loop, and the M relay follows. The RGR relay is a slow-release relay, assuring that the M relay will follow the A relay and not the RGD relay.

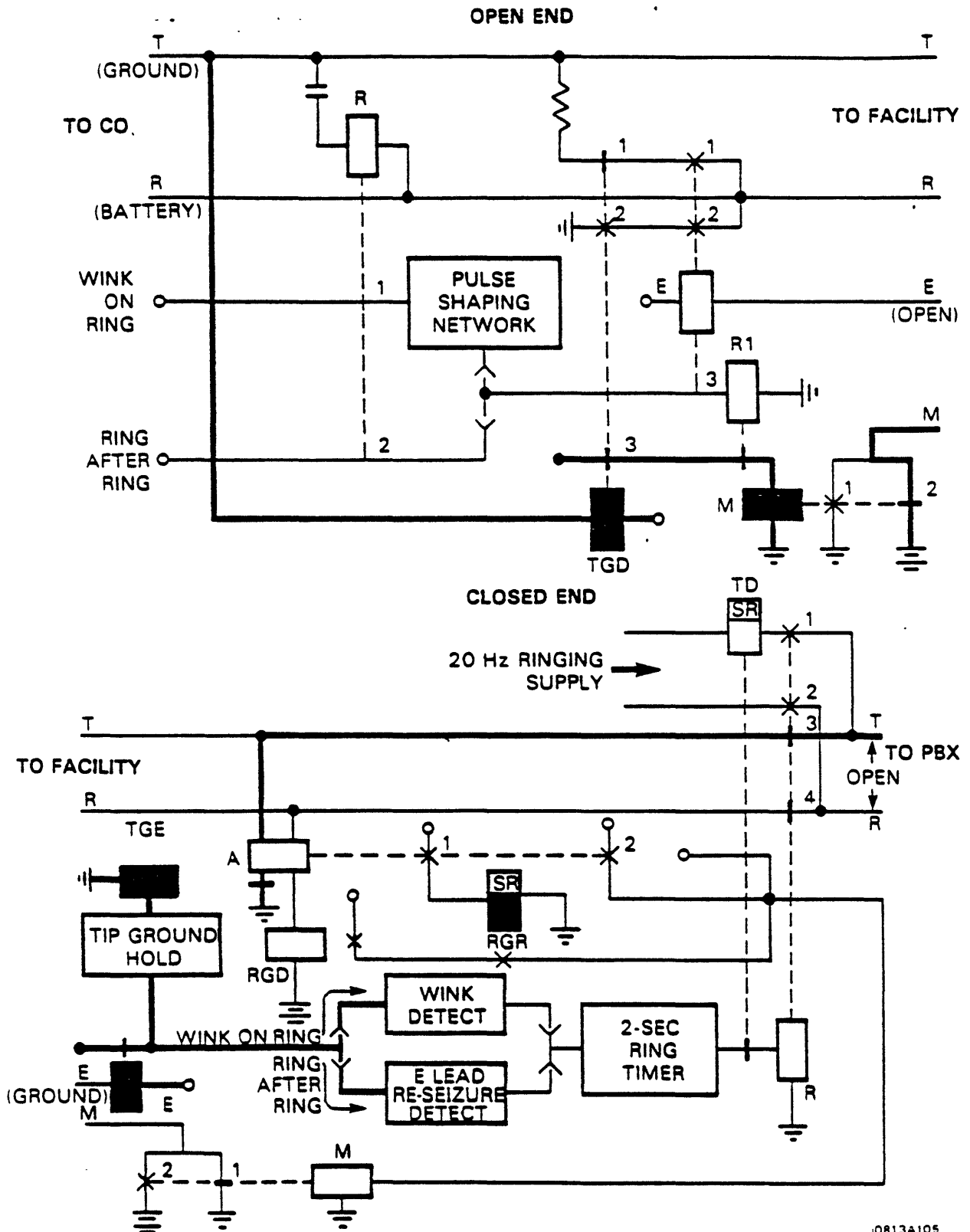


Figure B-5e. Dial Pulse Break

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At the open end, the E relay releases in response to the open on the E lead. The E(1) contacts open, opening the loop toward the CO, signaling the dial pulse break. (The E(2) contacts also open, but this has no effect because the TGD(2) contacts remain open.)

**Occupied Circuit:** Figure B-5f shows the circuit in the occupied condition.

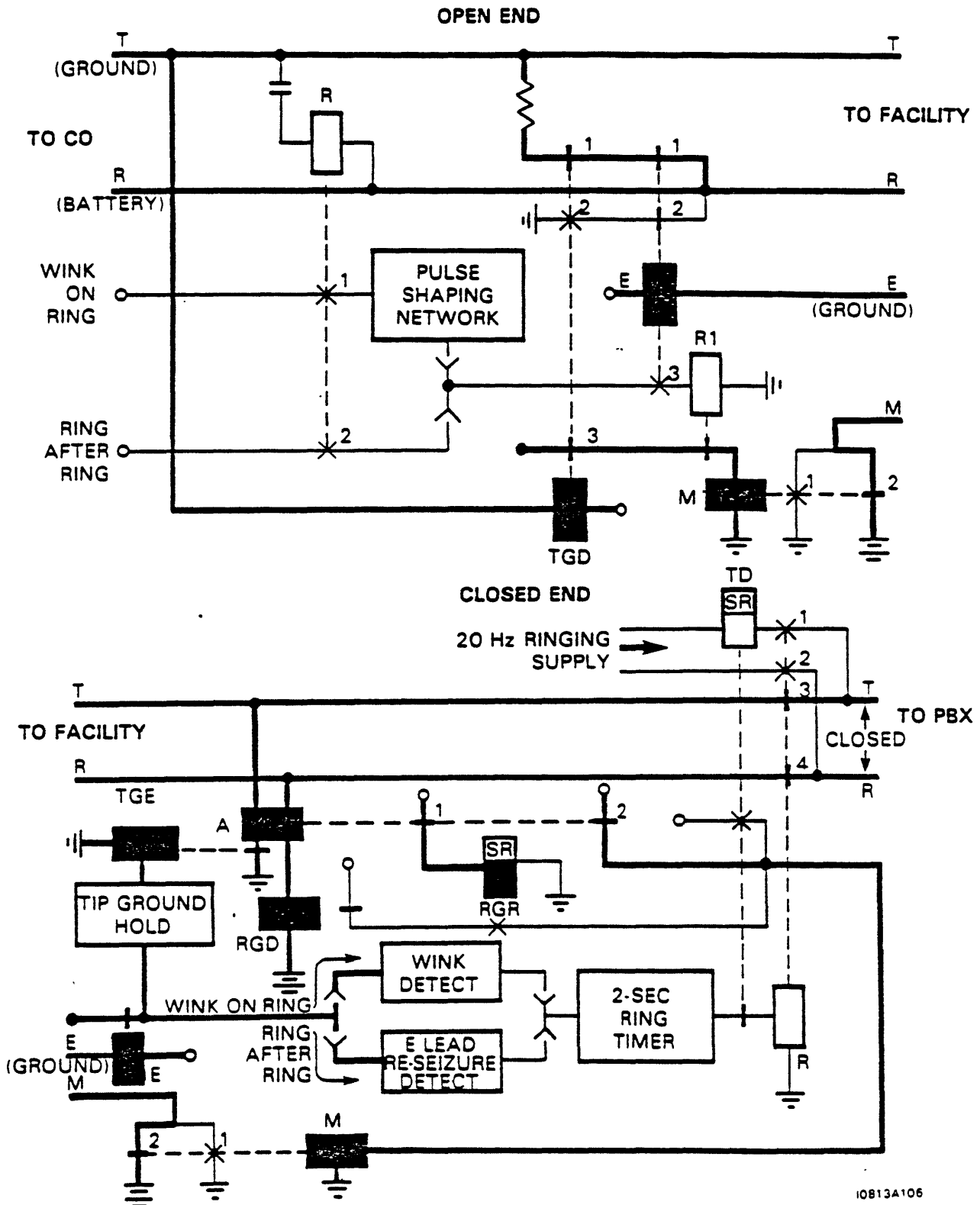


Figure B-5f. Occupied Circuit

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## Part I Trunking

**INCOMING CALL (CO ORIGINATION):** An incoming call is illustrated with reference to the following events and their associated figures:

EVENT	FIGURE
CO Seizure	B-6a
Incoming Ring	B-6b
Ring Tripping During Ring	B-6c
Answer During Rest and Occupied Circuit	B-6d

**CO Seizure:** The CO seizes the circuit by grounding the tip lead (Figure B-6a). The TGD relay operates, closing the TGD(3) contacts. The M relay operates, grounding the M lead toward the facility.

At the closed end, the E relay operates in response to ground on the E lead. The TGE relay operates, grounding the tip lead toward the PBX. The PBX recognizes the grounded tip as incoming seizure and will not seize the circuit for an outgoing call.

**Incoming Ring:** The CO applies 20 Hz ringing voltage, superimposed upon battery on the R lead (Figure B-6b). The R relay, which is sensitive to ac, operates, closing the R(1) and R(2) contacts.

In *wink-on-ring operation*, the pulse shaping network responds to the closure of the R(1) contacts with a brief pulse toward the R1 relay. The R1 relay operates in response to this pulse, and the M relay releases briefly, opening the M lead toward the facility. This is the "wink."

At the closed end, the E relay releases momentarily. The wink detect circuitry is designed to recognize the wink, and relays the wink to the 2-second ring timer. The ring timer supplies current to the R relay for 2 seconds. The R relay operates, closing the R(1) and R(2) contacts, applying ringing voltage toward the PBX. The R(3) and R(4) contacts open, preventing application of ringing voltage toward the facility.

In *ring-after-ring operation*, the R1 relay follows the R relay and is operated for the full ringing period, resulting in ground on the M lead.

At the closed end, the E lead is open for the full ringing period.

The E lead re-seizure detect circuit activates the 2-second ring timer *after* recognizing the 2-second on-hook followed by re-seizure. The R relay operates as in *wink-on-ring operation*, causing ringing voltage to be applied toward the PBX.

The *tip ground hold circuit* supplies current to keep the TGE lead from releasing during the 2-second open on the E lead. This is necessary to prevent the PBX from receiving a 2-second on-hook (open tip) signal prior to the 2-second ring.

**Ring Tripping During Ring:** If the call is answered during the ringing cycle, the ringing must be tripped immediately to prevent an annoying ring in the called party's ear. This is accomplished by tripping the ring at the near end.

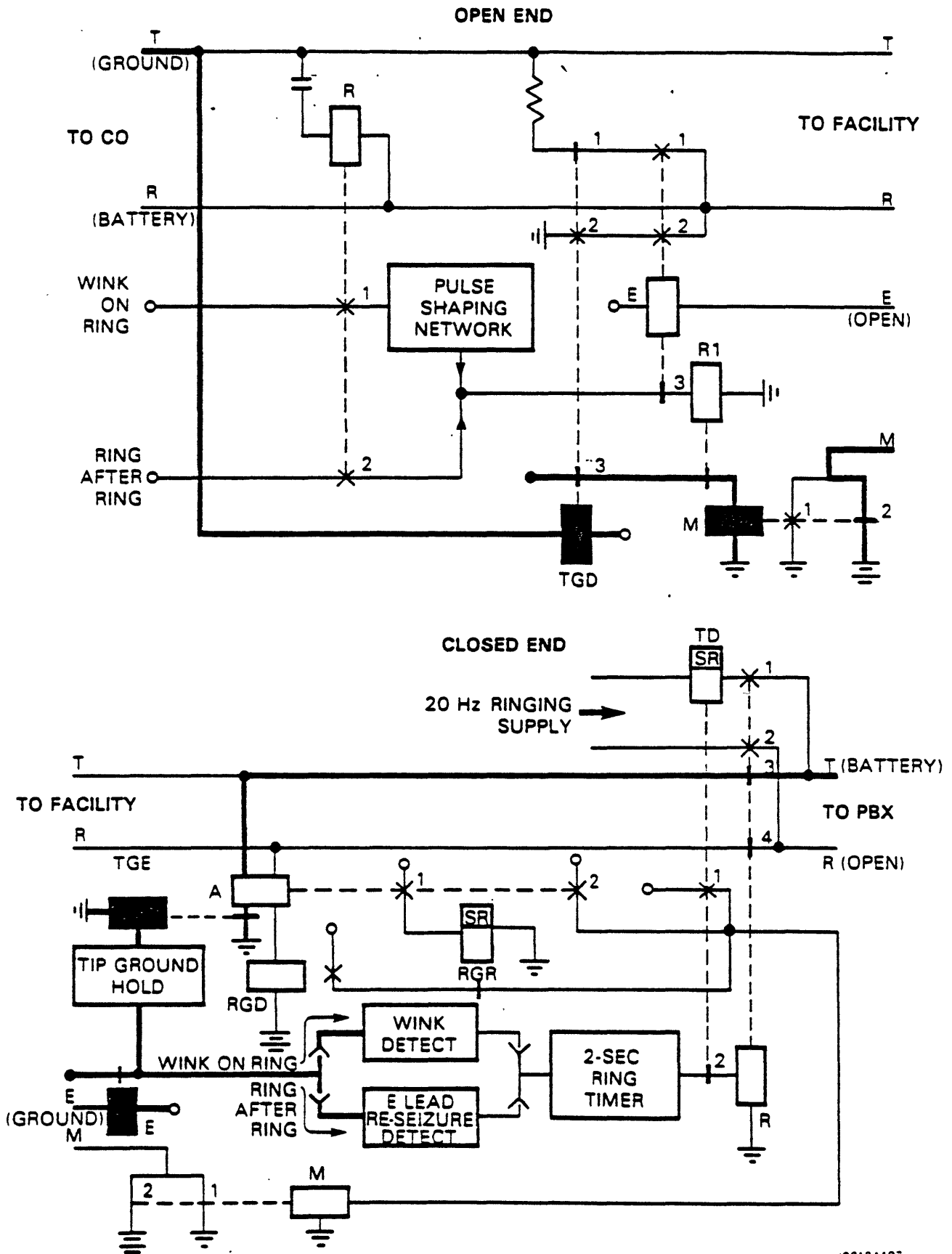
Direct current through the TD relay causes it to operate, opening the TD(2) contacts. The R relay releases, and the R(1) and R(2) contacts open, removing the ringing toward the PBX.

The R(3) and R(4) contacts open, and loop current operates the A and RGD relays, shown in transition in Figure B-6c.

The TD(1) contacts close, causing the M relay to operate, placing battery on the M lead toward the CO.

At the CO end, ground on the E lead operates the E relay, opening the E(3) contacts. The R1 relay releases, closing the R1 contacts. Current through the M relay causes it to operate (shown here in transition).

The E(1) contacts close, completing the loop toward the PBX. Once it recognizes the loop closure, the CO will remove the ringing voltage from the circuit, and the R relay will release (shown here in transition).



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Figure B-6a. CO Seizure

**Part I  
Trunking**

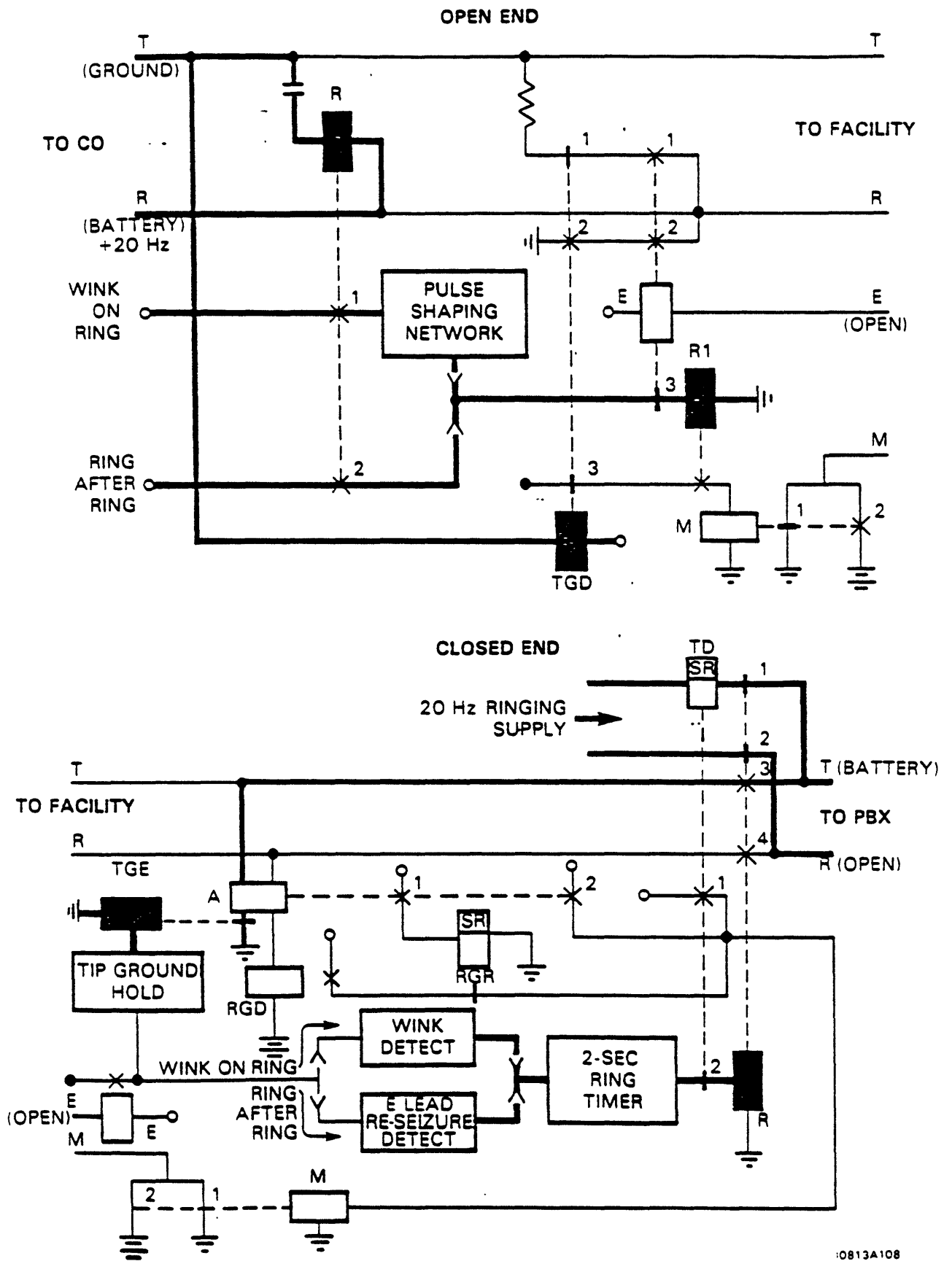
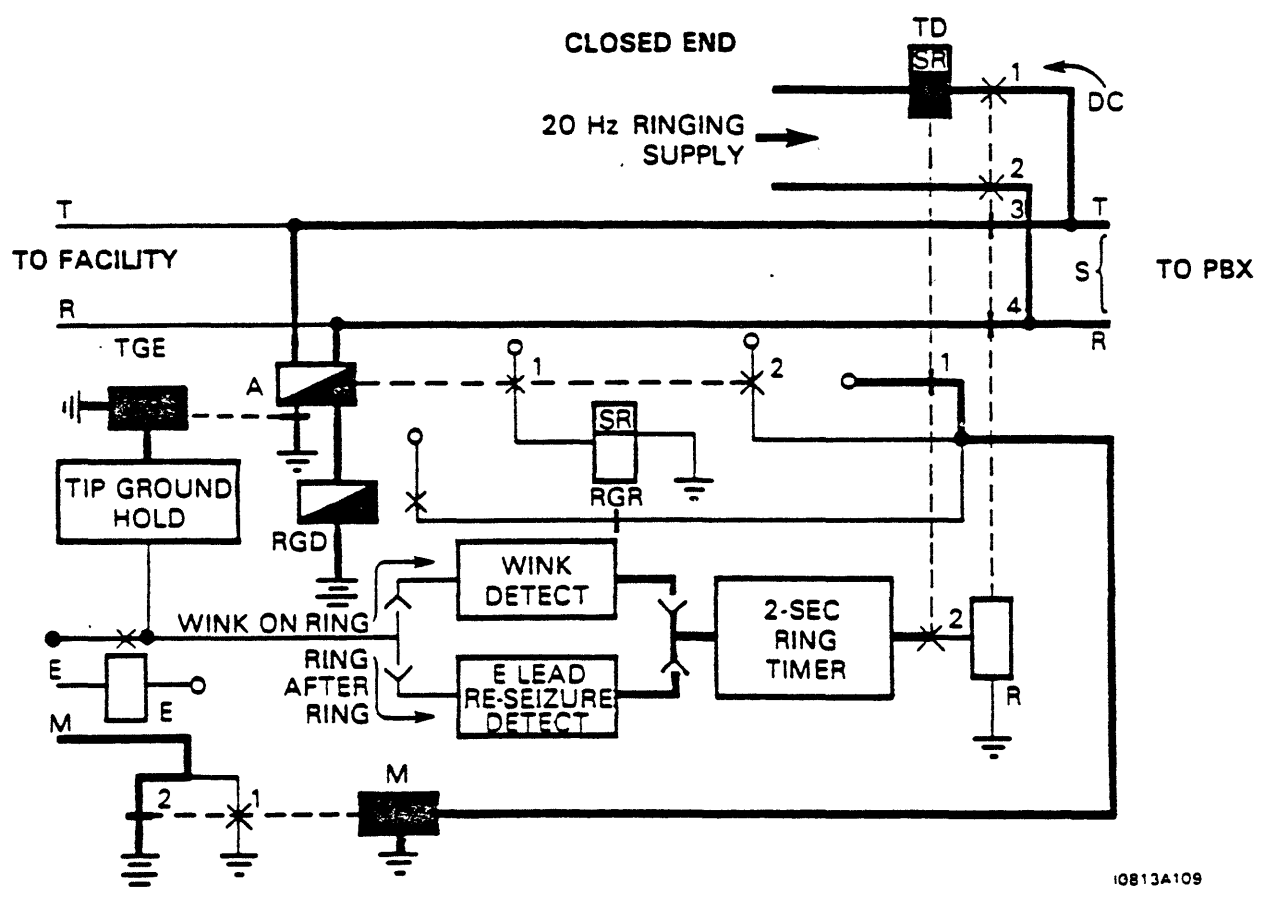
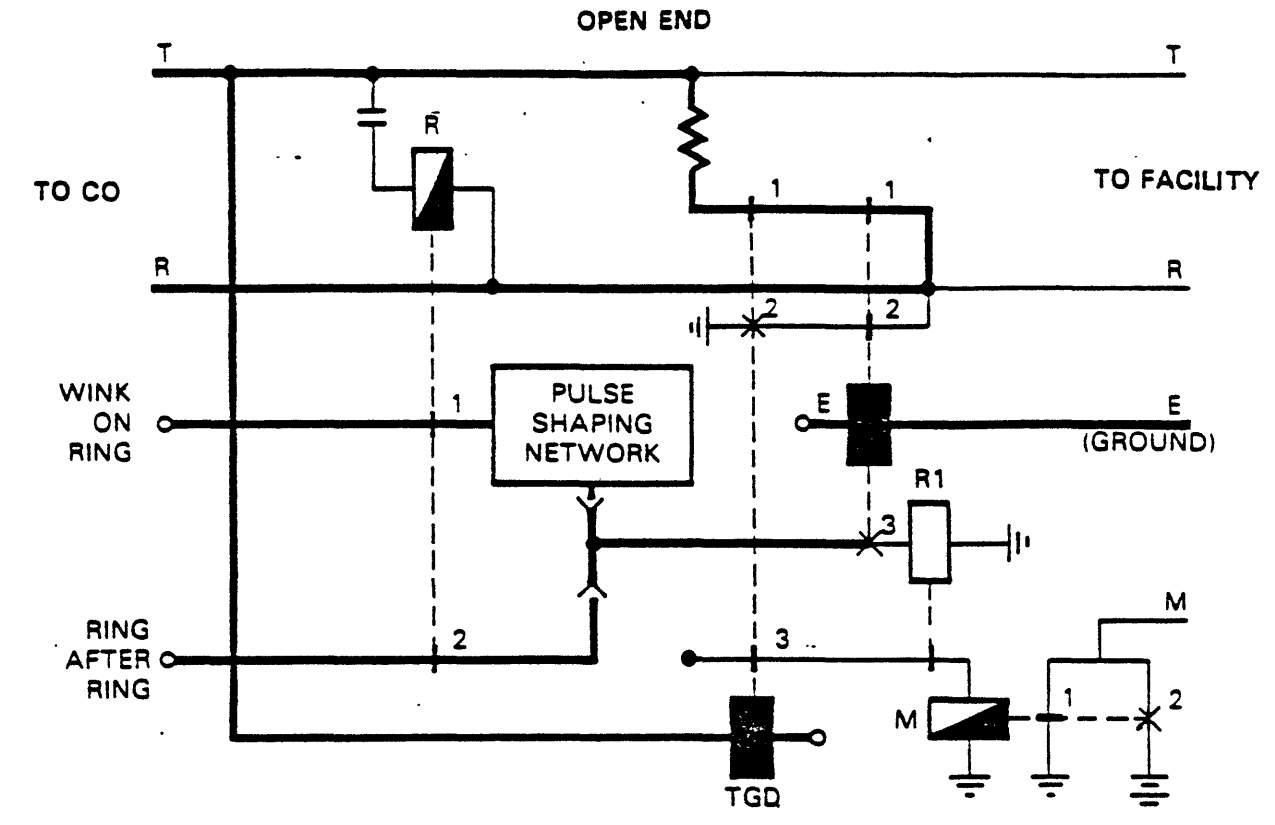


Figure B-6b. Incoming Ring



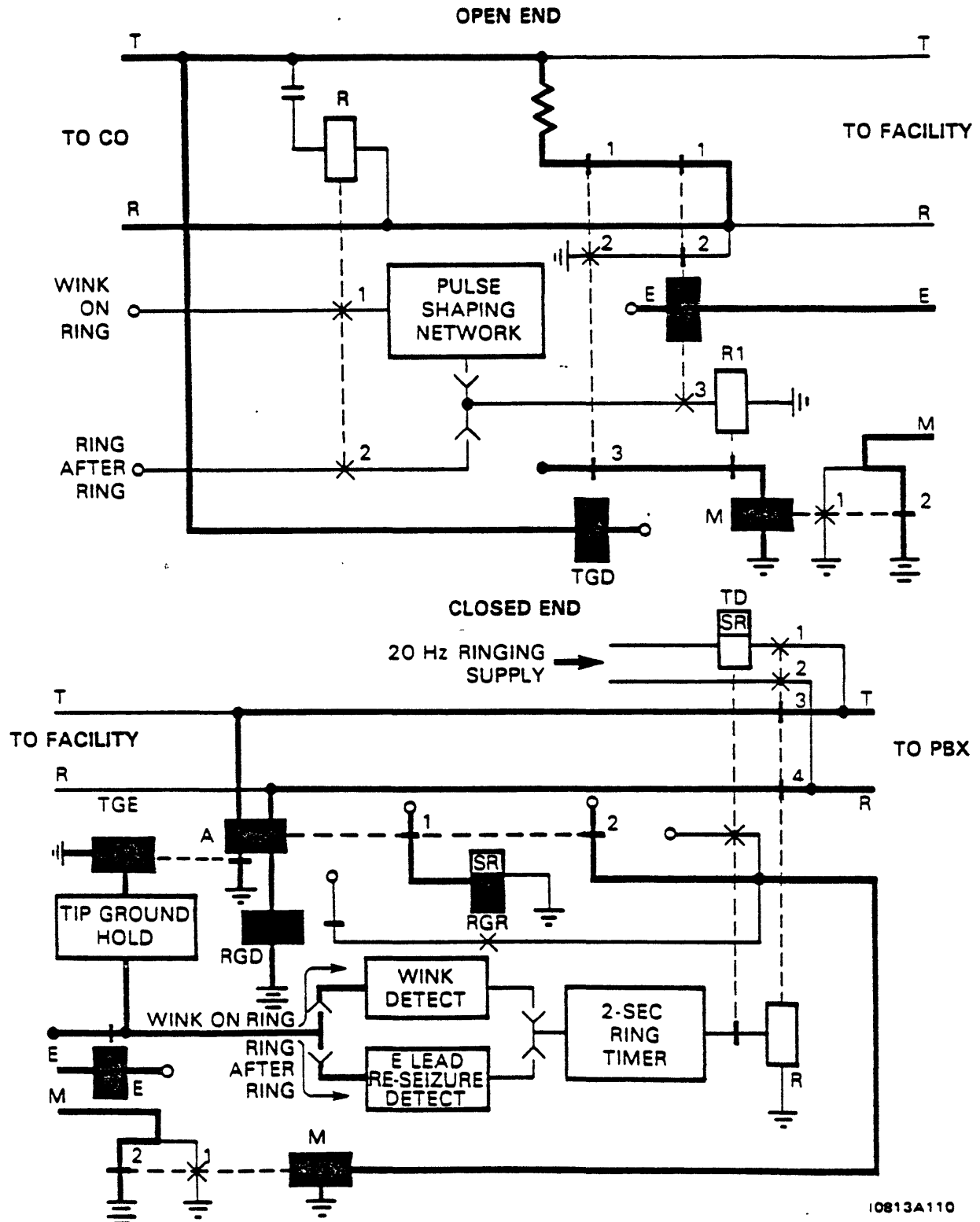
10813A109

Figure B-6c. Ring Tripping During Ring

## Part I Trunking

**Answer During Rest and Occupied Circuit:** Figure B-6d illustrates the completion of PBX answer after the ring has been tripped. It also illustrates PBX answer in the case where the PBX answers during the rest period between rings.

In the occupied state, the A relay is holding up the loop toward the PBX, and also is holding the M relay up via the closed A(2) contacts. At the CO end, the E relay is holding the loop toward the CO.



10813A110

Figure B-6d. Answer During Rest and Occupied Circuit



## APPENDIX C OVERALL DIRECT INWARD DIALING OPERATION









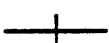
### OPERATION OF DIRECT INWARD DIALING (DID) SIGNALING

Overall DID operation will be discussed with reference to the simplified circuit schematics in Figures C-1a through C-1g. The events and their associated figures are:

EVENT	FIGURE
Idle Condition	C-1a
CO Seizure	C-1b
Start Dialing Wink	C-1c
Dial Pulsing	C-1d
PBX Answer	C-1e
PBX Disconnect	C-1f
CO Disconnect	C-1g

The symbols and conventions used in the circuit schematics are given in Table C-1.

Table C-1. Symbols and Conventions

SYMBOL	EXPLANATION
	Heavy line. Indicates active portion(s) of the circuit for the operation being described.
	Light line. Indicates inactive circuitry for the operation being described.
	Light relay symbol. No current in relay windings, relay released.
	Dark relay symbol. Current in relay windings, relay operated.
	Relay in transition.
	Relay momentarily operated.
	Relay contact closed.
	Relay contact open.
	Audio path.

## Part I Trunking

**IDLE CONDITION:** The idle condition is shown in Figure C-1a. All relays are released and the relay contacts are as shown.

In the idle state, tip and ring terminate in a high-resistance loop in the CO, represented here by capacitors in the tip and ring leads. No current flows in the RBD relay because the diode will only allow current from the tip to the ring. In other words, the diode will allow current to flow only with ground on the ring and battery on the tip in the PBX.

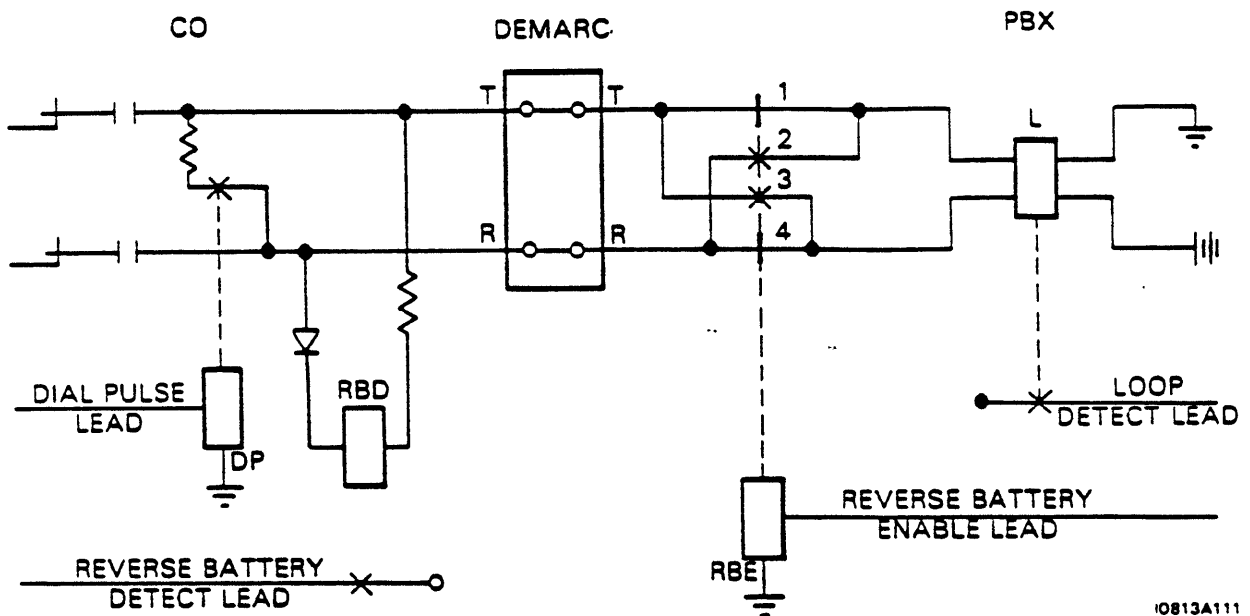


Figure C-1a. Idle Condition

**CO SEIZURE:** As mentioned earlier, DID trunks are one-way trunks and a call may be originated only from the CO end of the circuit (Figure C-1b). The CO seizes the trunk by operating the DP relay via the dial pulse lead. The DP contacts close, forming a low-resistance loop toward the PBX.

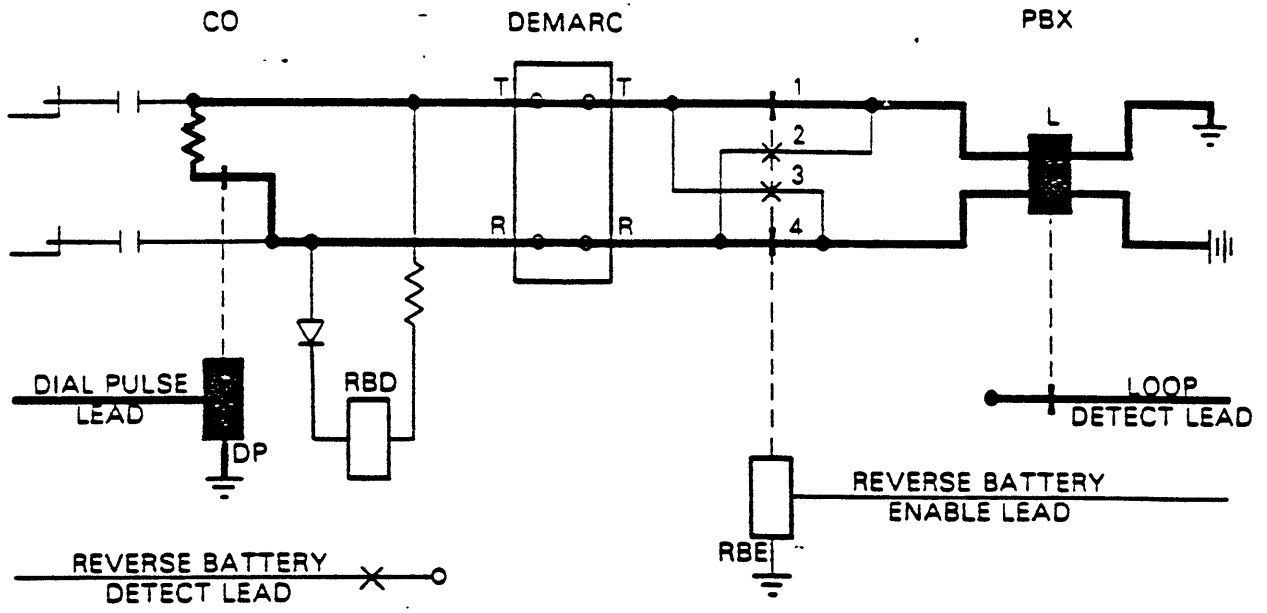
Current in the tip and ring leads operates the L relay in the PBX, closing the L contacts. The PBX recognizes the seizure via the loop detect lead.

Although the voltage across the diode in the RBD circuit decreases, the tip lead remains at a higher potential than the ring lead and no current flows through the RBD diode.

**START DIALING:** Upon recognizing the seizure, the PBX attaches digit collection equipment to the circuit. Once it is prepared to receive digits, the PBX sends a start dialing signal to the CO (Figure C-1c). The start dialing signal consists of a brief reversal (a "wink") of battery and ground toward the CO.

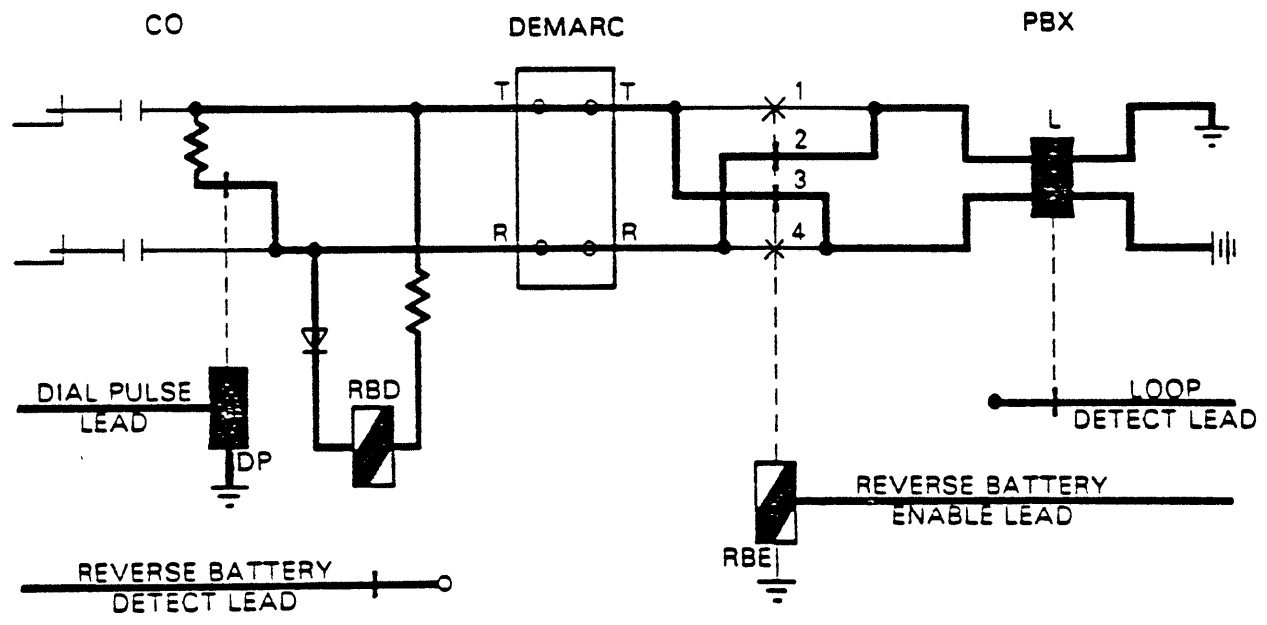
During the wink, the PBX operates the RBE relay via the reverse battery enable lead. Contacts RBE(1) and RBE(4) open, and contacts RBE(2) and RBE(3) close.

At the CO end, battery reversal causes the diode to operate. Current flows in the RBD relay, operating the relay. The CO recognizes the reverse battery via the reverse battery detect lead.



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**Figure C-1b. CO Seizure**



10813A113

**Figure C-1c. Start Dialing "Wink"**

**Part I  
Trunking**

**DIAL PULSING:** Upon recognizing the reverse battery "wink," the CO proceeds to send dial pulse address digits (Figure C-1d).

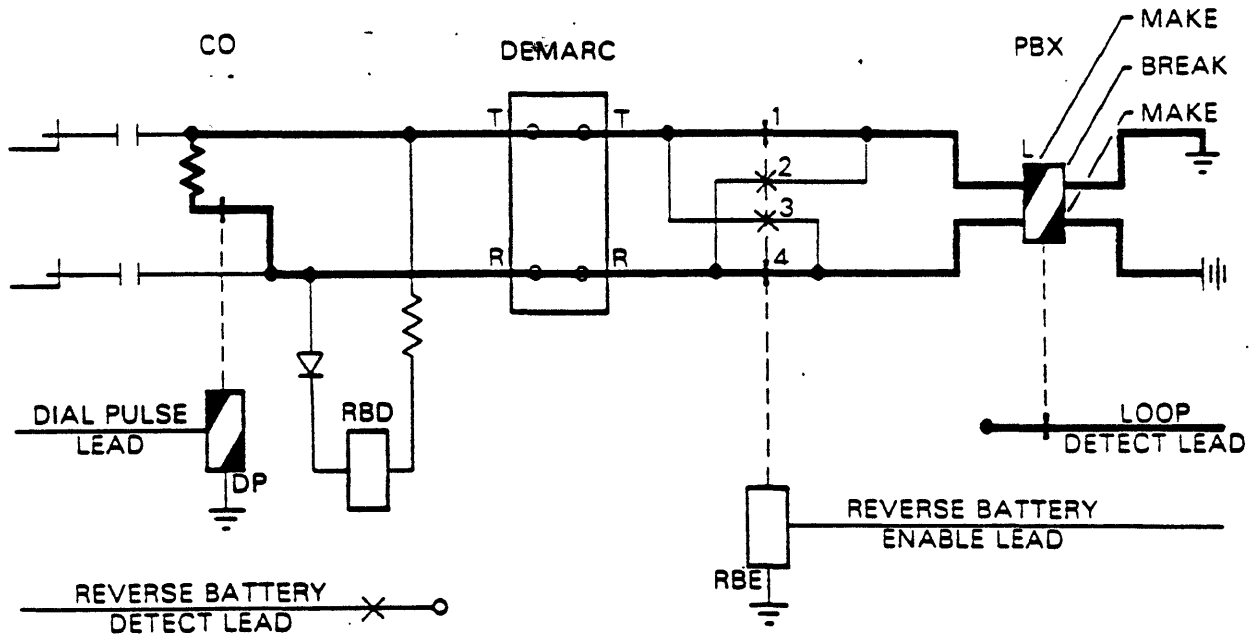


Figure C-1d. Dial Pulsing

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**PBX ANSWER:** The PBX routes the call and rings the extension (Figure C-1e). When the extension answers, the PBX again reverses the battery by operating the RBE relay. The CO recognizes the reversal as answer supervision.

The battery remains reversed for the duration of the call.

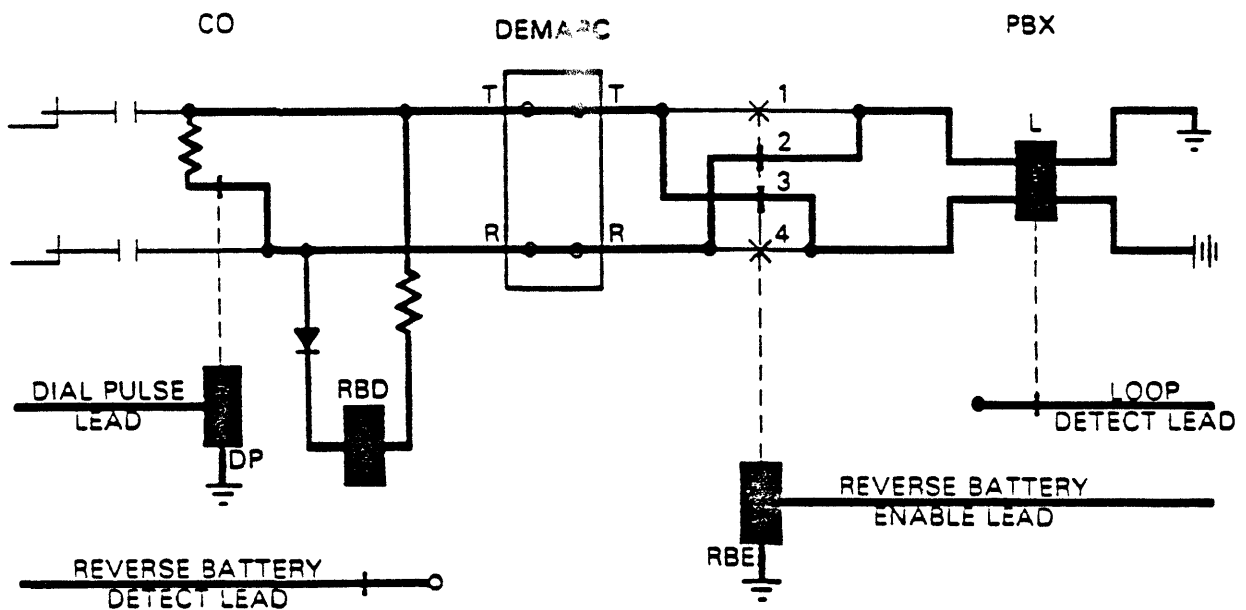


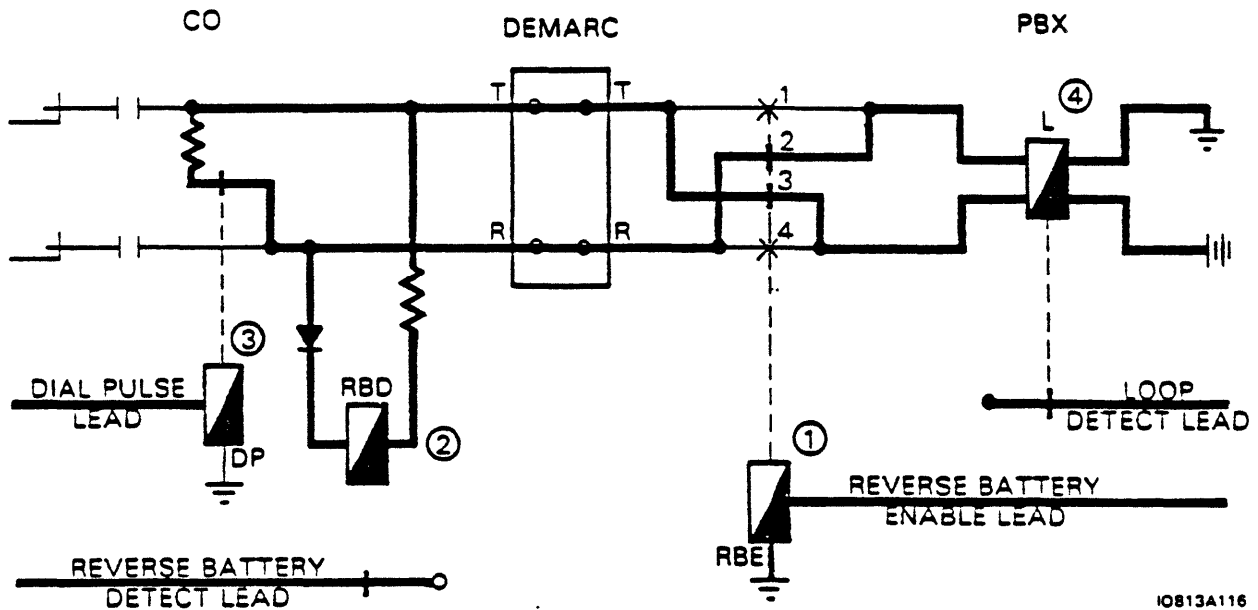
Figure C-1e. PBX Answer

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**DISCONNECT:** Either end may initiate a disconnect.

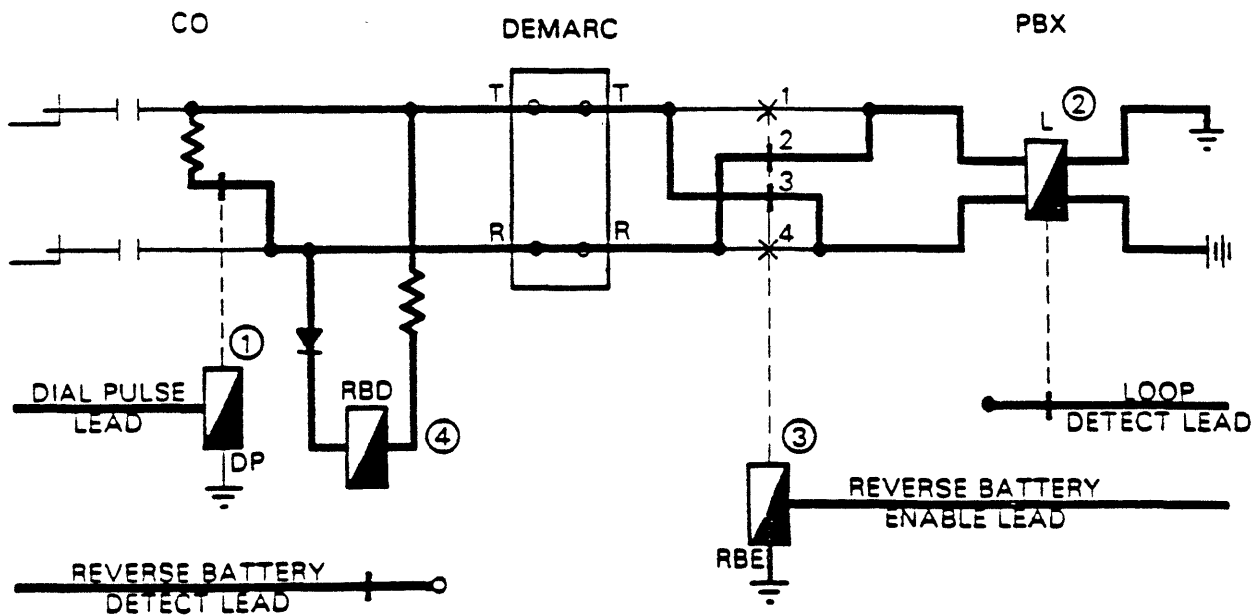
If the PBX extension goes on-hook first, the PBX releases the RBE relay, and the RBD relay in the CO follows. The CO recognizes the disconnect via the reverse battery detect lead and releases the DP relay, opening the low-resistance loop. The L relay in the CO releases and the circuit returns to the idle condition. In Figure C-1f, the relays are shown in transition. The sequence of operation of the relays is shown in the circled numbers.

If the distant end goes on-hook first, the CO releases the DP relay and the L relay in the PBX follows (Figure C-1g). The PBX releases the RBE relay, and the RBD relay in the CO follows, returning the circuit to the idle condition.



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**Figure C-1f. PBX Disconnect**



IOB13A117

**Figure C-1g. CO Disconnect**



## **APPENDIX D ANSWERS TO REVIEW QUESTIONS**

### **CHAPTER 1**

1. Switching, signaling and power equipment may be centralized and shared by all users.
2. Line circuits are dedicated to a single station set; trunk circuits connect switching systems and are shared among users.
3. Specialized functions, convenience, economy and cost control.
4. To connect private networks to the public switched network.
5. FX trunks provide a direct connection between local customer equipment and a distant CO.
6. In-WATS calls are billed to the called number.
7. Tie.
8. Inward calls do not have to be served by an attendant.
9. To provide the convenience of extension dialing to remotely located telephones
10. The exchange of control information.
11. On-hook and off-hook.
12.  $\text{Percent Break} = 100 \times \frac{\text{Break duration}}{\text{Pulsing period}}$
13. 16
14. Signaling in CCIS is carried over separate signaling links.
15. Dial tone, audible ring, busy, reorder, recorded announcements.
16. Senderized and non-senderized.
17. Origination, routing, answer supervision, disconnect, and clean-up.
18. Reorder is returned to the calling party.

### **CHAPTER 2**

1. Metallic facilities are capable of carrying direct current; carrier facilities are not.
2. T and R toward facility; T1 and R1 from facility.
3. Direct current and tone.
4. To convert from 2-wire to 4-wire transmission.
5. Glare is the simultaneous seizure of both ends of a trunk.
6. Wink start and delay dialing

## Part I Trunking

### CHAPTER 3

1. Glare. No positive control of outpulsing digits from the PBX to the CO. No positive signaling to PBX of distant-end disconnect.
2. 

	Tip	Ring
CO	ground	battery
PBX	open	open
3. By closing the loop toward the CO.
4. The CO applies interrupted ringing voltage, superimposed on  $-48$  Vdc, toward the PBX.
5. If the carrier path is interrupted, it will be interpreted by the E&M to SF unit as an incoming ring. This may be avoided by using inverted E&M lead operation.

### CHAPTER 4

1. 

	Tip	Ring
CO	open	battery
PBX	battery	open
2. Grounds the tip, attaches a digit receiver and sends dial tone to the PBX.
3. Tip ground from the CO provides the PBX with a positive start dialing signal, reduces glare incidences, and provides a positive signal indicating that the CO has disconnected.
4. Grounds the tip lead toward the PBX.
5. The PBX recognizes the tip ground from the CO as incoming seizure and will not seize the circuit for an outgoing call. Glare is minimized.
6. Removes the tip ground.
7. The open-end MFT sends a B/G/B wink toward the AFT over the M lead.
8. The M lead toward the AFT is grounded for the full ringing period.

### CHAPTER 5

1. Controlled outpulsing provides terminating equipment with a method of informing the originating switch that it is prepared to receive address digits.
2. CO transition from off-hook to on-hook.
3. An on-hook to off-hook to on-hook "wink."
4. Wink start

### CHAPTER 6

1. DID allows an off-net caller to directly dial an extension served by the PBX, without being served by an attendant.
2. 

	Tip	Ring
CO	high	resistance loop
PBX	ground	battery
3. No DID trunks are *one-way incoming*.
4. By changing the loop from high to low resistance
5. A brief tip-ring reversal, or "wink"
6. By reversing the state of the tip and ring leads. The tip is connected to battery and the ring is grounded



**PART II**  
**TRANSMISSION**



## **INTRODUCTION TO TRANSMISSION**

The field of communications deals with the interchange of thoughts and ideas between and among individuals and groups.

Most human communication involves the exchange of information in audible form or in visual form. The most common forms of audible communication are speech and music. Visual communication is conducted via pictures and the written word.

The field of telecommunications is concerned with carrying information by means of electrical energy flowing over wires or radiated through space. The use of electrical energy allows virtually instantaneous communication over distances that are, for all practical purposes, unlimited.

Many different types of telecommunications systems have been developed. Those most familiar to us in our everyday lives are radio, television and the telephone system. This text is concerned with the underlying principles of voice communication over the telephone system.

The telephone system accomplishes the transmission of speech by converting the sound energy of the speaker's voice into electrical energy that has similar characteristics. The electrical energy is carried over wires and radiated through space to the desired destination, and it is then converted back into sound energy at the listener's station.

The quality of a telephone transmission system depends upon how close it duplicates the spoken voice at one end with the reproduced voice at the other end. Anyone who has used the telephone has experienced "good" and "bad" connections and could probably describe the quality of a particular connection in a subjective way. The purpose of transmission engineering is to establish *objective* and *measurable* standards that, when met, will ensure that the user of the system will experience "good" connections.

The purpose of transmission testing is to ensure that a circuit or connection is performing as it was designed to perform.

Chapter 1 reviews the characteristics of human speech and hearing as they are related to telephone transmission.

Chapter 2 defines the units of measurement that are used in transmission testing.

Chapter 3 contains a general discussion of transmission parameters that must be controlled in order to avoid impaired transmission.

Chapter 4 discusses the interfaces between customer-provided equipment and telephone company-provided facilities.

Chapter 5 contains a detailed discussion of transmission testing.



## CHAPTER 1

# CHARACTERISTICS OF HUMAN SPEECH AND HEARING

### INTRODUCTION

The transmission of speech to a remote point by electrical means involves three essential processes:

1. Converting speech sounds to an electrical signal which varies in a manner similar to the sound.
2. Passing the electrical signal over connected transmission media until the receiving point is reached.
3. Converting the electrical signal back into sound that closely duplicates the original speech.

The subject of transmission deals with 2 above. However, prior to discussing the various aspects of transmission, it is helpful to review the basic characteristics of sound, speech and hearing, with the objective of understanding *what it is* that is converted to and from electrical signals in a transmission system.

### SOUND AND SOUND WAVES

Everyone is familiar with the series of ripples that spread from a stone cast upon the still waters of a lake or pond. The "ripples" are one of many forms of wave motion and carry energy away from the impact of the stone by alternately raising (the crest) and lowering (the trough) the surface of the water.

Sound is similar to the ripples on a pond. It is a form of wave motion produced by some vibrating body such as a bell, a tuning fork or the human vocal cords. The "waves" consist of alternating compressions (high pressure, corresponding to the crest of the water wave) and rarefactions (low pressure, corresponding to the trough of the water wave).

Water waves radiate from a single point in a series of rings. Sound waves, on the other hand, are a series of concentric spheres expanding at a definite rate of travel. This rate of travel (or the *velocity* of the wave) is approximately 1075 feet per second, varying with atmospheric conditions and altitude.

Figure 1-1 shows a vibrating tuning fork radiating sound waves. Although the sound waves are radiated in three dimensions, it is convenient to represent the waves in a two-dimensional graph, as in the bottom of the figure.

The important quantities in describing this wave, and in fact any wave, are shown in the diagram.

The height or *amplitude* of the wave is the vertical distance between the trough and the crest. In the case of sound waves, the amplitude is measured in terms of pressure. (The actual values are not shown here, as they are not important to a basic understanding of sound waves.)

The *period* of the wave is the time, measured in seconds, it takes a full wave to pass a fixed point, such as A.

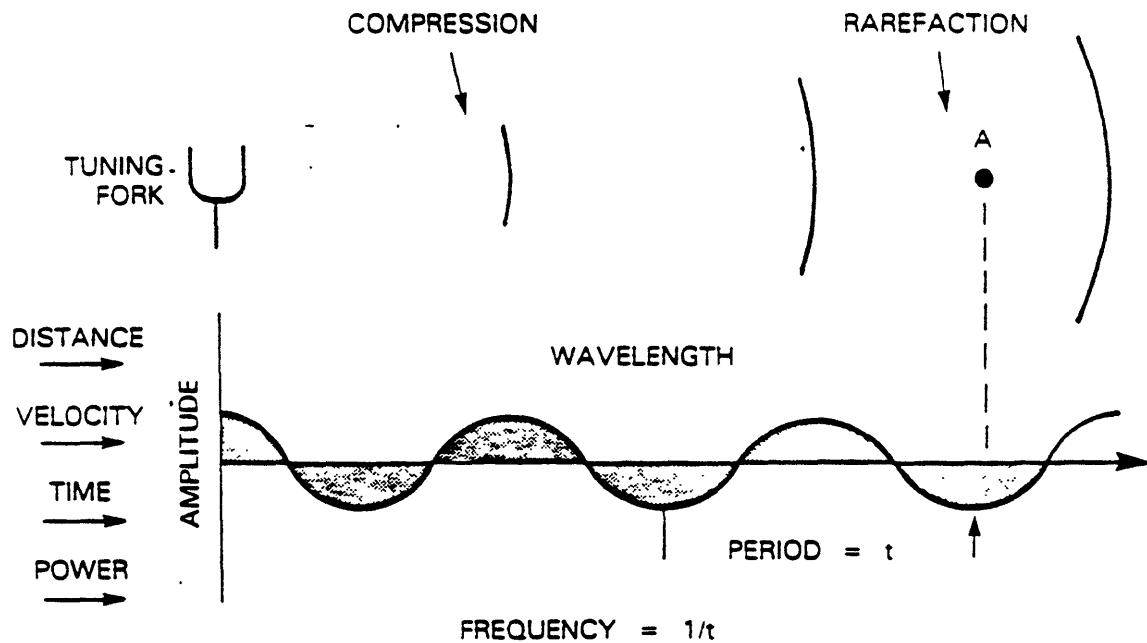
The *frequency* is the number of times the tuning fork vibrates per second. An equivalent definition of frequency is the number of waves that pass an arbitrary point, say A, per second. The units of frequency are *cycles per second*, and one cycle per second is called one *hertz*, abbreviated *Hz*.

There is a simple relationship between period and frequency.

$$\text{frequency} = 1/\text{period}$$

The *wavelength* is the distance between successive wave crests, measured in units of length.

**Part II  
Transmission**



10813A118

**Figure 1-1. Sound Waves**

The *velocity* of the wave, in this case 1075 feet per second, is the speed with which the wave travels through the air.

For any wave, there is a simple relationship between frequency, wavelength and velocity.

$$\text{velocity} = \text{frequency} \times \text{wavelength}$$

For example, the tuning fork in Figure 1-1 is vibrating at 460 Hz. What is the wavelength?

Re-arranging the equation,

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}}$$

or

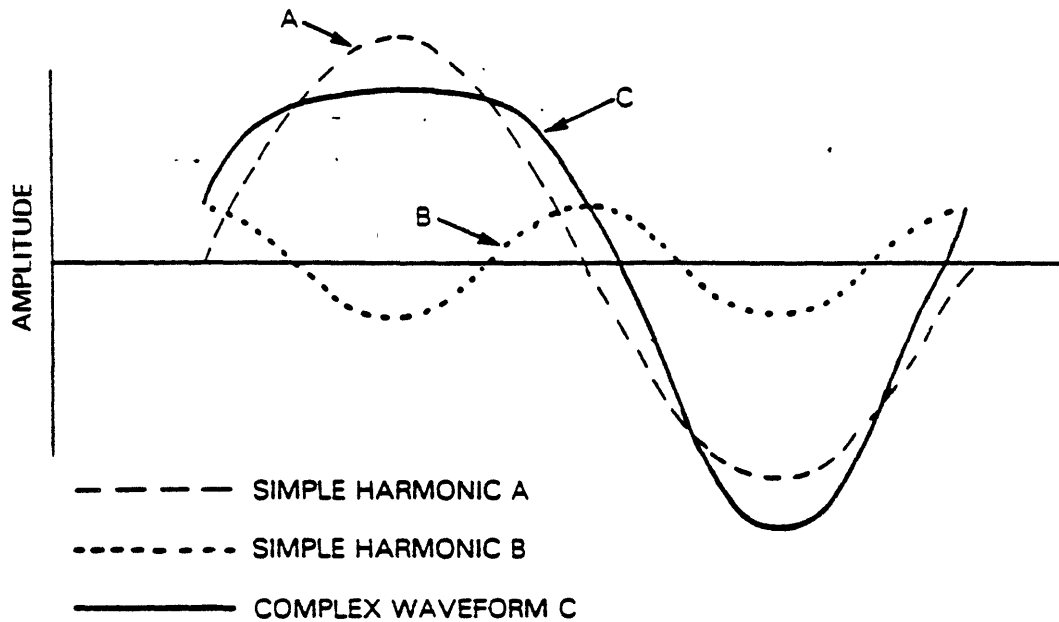
$$\begin{aligned} \text{wavelength} &= \frac{1075 \text{ ft/sec}}{460 \text{ cycles/sec}} \\ &= 2.33 \text{ ft} \end{aligned}$$

The *energy* present in the wave is directly related to the shaded area in Figure 1-1. The energy carried by a wave increases as the amplitude of the wave increases.

The *power* is defined as energy per unit of time. In Figure 1-1, the power at point A is directly related to the shaded area that passes point A per second. The power contained in sound waves may be described in units of *watts*—the same units that are used in describing electrical power

Power, like energy, increases as the size of the wave increases. For sound waves, the louder the sound, the greater the power.

Of the quantities that may be used to describe sound (and, later in the text, electrical waves), frequency and power are the most useful.



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**Figure 1-2. Formation of Complex Waves**

## **PURE AND COMPLEX SOUNDS**

If the source of a sound is a simple mechanism such as a tuning fork, it will produce a wave that varies from maximum pressure (the crest) to minimum pressure (the trough) in a smooth manner. Wave motion of this type is called *simple harmonic*, and is represented by a *sine curve*, as in Figure 1-2. Sounds that contain a single simple harmonic are called *pure tones*.

If a sound is produced by a mechanically complex device, such as a musical instrument or the human vocal cords, it will contain several simple harmonics which add together to form a complex sound.

It is characteristic of simple harmonics, or sine waves, that they may be added together in a straightforward manner to produce complex sounds.

For example, take the two simple harmonics, A and B, in Figure 1-2. They may be combined to form the complex waveform, C, by adding their amplitudes at all points.

The waveform C in Figure 1-2 has a different frequency and amplitude than either A or B, and it also has a different shape; that is, it is no longer a sine wave.

By successively adding waveforms in the manner that we added A and B, we can come up with waveforms that are of any complexity we desire. In fact, it is possible to construct any desired waveshape by adding together the appropriate simple harmonics.

An equivalent way of stating this is that complex waveforms are combinations of simple waveforms. In fact, it can be shown that any complex periodic waveform found in nature is composed of a number of simple waveforms. It can also be shown that these component waveforms can be located and measured.

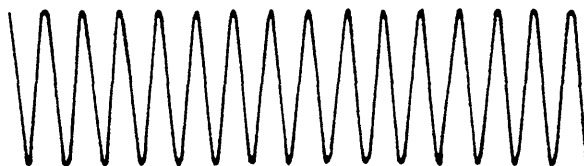
To summarize,

- Any periodic waveform in nature is composed of a number of simple harmonic waveforms that may be physically identified and measured. The component waveforms can be completely described in terms of their *power* and *frequency*
- The *total power* carried by a complex waveform is equal to the sum of powers contained in each component simple harmonic waveform
- This applies to both sound waves and electrical waves

**SPEECH**

Speech sounds are produced by the combined action of the lungs, the vocal cords and the air passages of the throat, mouth, and nose.

Speech sounds are, in general, complex sounds. The sounds contained in even the simplest syllable are composed of several harmonics which add together to form a complex waveform. This is illustrated in Figure 1-3, which includes graphs of the vowel sounds  $\bar{o}$  and  $\bar{oo}$ .



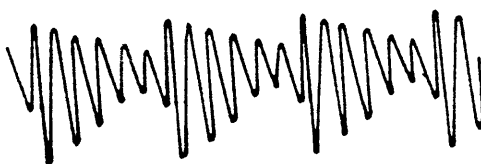
SIMPLE SOUND



$\bar{oo}$  AS IN LOOSE



$\bar{o}$  AS IN LOW



MUSICAL NOTE



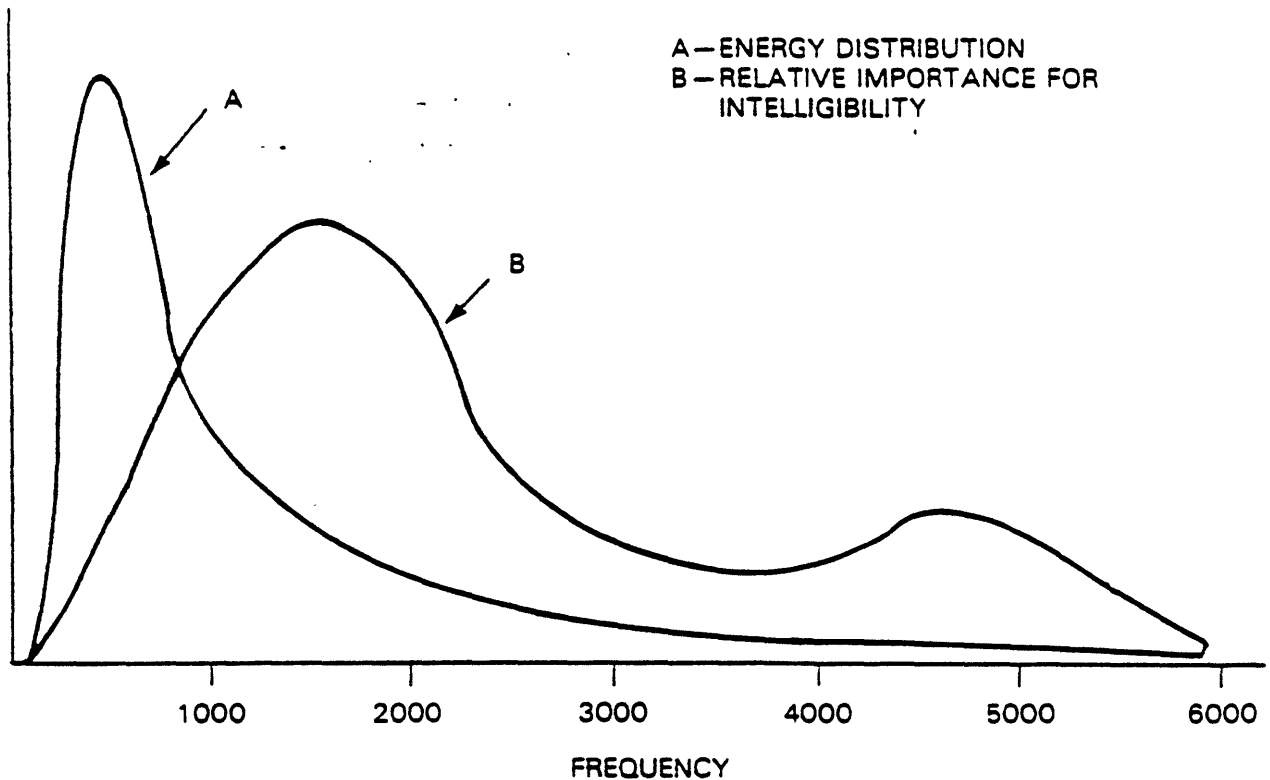
NOISE

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**Figure 1-3. Sound Waveforms**

Because most speech sounds are complex and there are hundreds of discrete speech sounds possible, it makes little sense to try to graph the component waveforms. A more useful and convenient way to characterize speech is to plot a graph of frequency versus power for the full voice range, as in Figure 1-4.





10813A121

Figure 1-4. Voice Frequency Range

Curve A shows that most of the energy contained in speech is concentrated between 0 and 2000 Hz. Although the frequencies above 2000 Hz contribute little to the overall energy contained in speech sound, the frequencies between 2000 and 6000 Hz are important to the clarity and intelligibility of sound. This is illustrated by curve B. The normal range of human voice is from 0 to 6000 Hz.

When compared to everyday notions about power, the power transported by speech waves is extremely small and is usually expressed in microwatts (abbreviated  $\mu\text{W}$ ). The average power for average talkers is about  $10 \mu\text{W}$  (that is, ten millionths of a watt!). Speech energy is also extremely variable. The softest sound made by a quiet talker contains approximately  $.001 \mu\text{W}$  of power; the loudest speech sounds contain about  $10 \text{ mW}$  (milliwatts) of power

## HEARING

The ability of the ear to hear a particular sound depends upon the frequency (*pitch*) and amplitude (*loudness*) of the sound.

Figure 1-5 is a graphical illustration of the range of human hearing. The "Threshold of Audibility" is the loudness at which a particular sound is barely perceptible. The graph shows the frequency range of the ear to be from about 20 to 20,000 Hz, with the greatest sensitivity to sound being in the range from about 500 to 4000 Hz. (The sensitivity is measured in decibels; a further discussion of decibels is contained in Chapter 2.)

## Part II Transmission

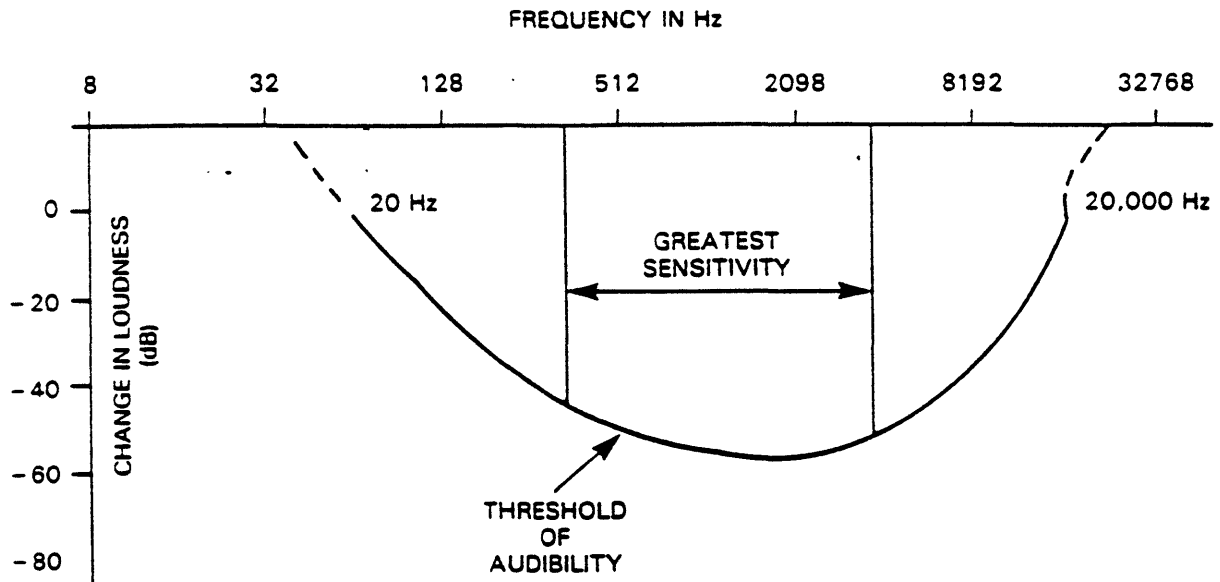


Figure 1-5. Frequency Range of Hearing

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### VOICE FREQUENCY BAND

To reproduce sounds across the full range of human hearing, telephone systems would have to have a frequency range of 20 to 20,000 Hz. This would be extremely expensive, and experience has shown it to be unnecessary. The *bandwidth* of telephone systems is from about 300 to 3000 Hz.

## **CHAPTER 1 REVIEW QUESTIONS**

1. What are the most useful quantities used in describing waveforms?
2. What is the term used to describe a pure tone, and how is it represented graphically (or mathematically)?
3. How are complex waves formed?
4. What is the frequency range of the human voice?
5. What portion of the frequency range contains most of the speech energy?
6. What portion of the voice frequency range is most important for intelligibility?
7. What is the transmission bandwidth of telephone circuits?



## CHAPTER 2 UNITS OF MEASUREMENT IN TRANSMISSION

### INTRODUCTION

The “quality” of a transmission system depends on “how close” it duplicates spoken voice at one end with reproduced voice at the other end. Anyone who has used the telephone has experienced both “good” and “bad” connections and could probably describe the quality of a particular connection in a subjective way.

The obvious question here is: What is meant by “quality,” “how close,” “good” and “bad”? In transmission, the first step in answering this question is to decide upon the following:

- What is to be measured
- What are the measurement units
- What is the reference point for the measurements

Any waveform may be characterized in terms of frequencies and power. It is not surprising, then, that the quantities commonly used to describe various aspects of transmission performance are *frequency* and *power*. Many performance standards are stated in terms of power at a particular frequency

The unit used to measure frequency is the *hertz*, abbreviated *Hz*.

As is common in most electrical systems, power is measured in units of *watts*, abbreviated *W*. Since the power encountered in transmission systems is relatively small (compared, say, to the power of a light bulb), power is usually expressed in *milliwatts*, abbreviated *mW*.

$$1 \text{ mW} = \frac{1}{1000} \text{ W} = .001 \text{ W} = 10^{-3} \text{ W}$$

In transmission, we are commonly interested in power ratios rather than in absolute power. In addition, transmission is concerned with an extremely wide range of absolute power values. For these reasons, a convenient mathematical expression of *relative power*, the *decibel* (abbreviated *dB*), is commonly used.

To describe relative power in terms of decibels, the reference point from which we are measuring must be defined. Depending upon the transmission parameter being measured, different forms of decibel measurement may be used. Each form of measurement has a specifically defined reference point.

Using the appropriate units of power related to specific references, we may measure absolute power, relative power and power gains and losses.

### MILLIWATT AND HERTZ

Since the power in telephone circuits is small, the milliwatt is used as the basic power measurement unit, just as the foot is used as the basic measurement of length in building a house.

Most measurements of *absolute power* in transmission are made in milliwatts or in units that are directly related to milliwatts.

The frequencies that are used in testing usually fall within the voice frequency band. Commonly used pure (sine wave) test tones are 404 Hz, 1004 Hz and 2804 Hz. (The 4 Hz offset isn't always stated; however, actual test frequencies should be offset by 4 Hz to compensate for effects that some carrier facilities have on test tones.) A measurement of 1004 Hz is near the voice band frequencies that carry much of voice power; 404 Hz is near the low end of the spectrum; and 2804 Hz is in the range of higher frequency components of the voice spectrum that are important to the intelligibility of speech.

## Part II Transmission

In addition to pure test tones, "white noise" within specific frequency ranges is used for certain tests. White noise test tones are complex waveforms that have their power evenly distributed over the frequency range of interest.

An example is useful to illustrate, in a very general and simplified way, how a transmission test is set up and how test tones are generated and measured.

Figure 2-1 illustrates the circuit that is under test. The equipment is set up to test the circuit between the demarc at A and the demarc at B. We are going to measure 1004 Hz *loss* inherent in the circuit between A and B.

The bridging clips at both demarcs are removed in order to isolate the segment of the circuit under test. At A, an oscillator is attached to the tip and ring leads. At B, a *Transmission Measuring Set (TMS)* is attached to the tip and ring leads.

The oscillator at A is set to generate a pure test tone with a power of 1 mW at 1004 Hz. At demarc B, the TMS is set to read power in the range of 1 mW. The power reading at B is 0.5 mW

The power lost between A and B is therefore  $1 \text{ mW} - 0.5 \text{ mW} = 0.5 \text{ mW}$ .

A more useful way of expressing the loss is in terms of the *relative loss*, or the ratio between power out (B) and power in (A).

$$\text{Relative Loss} = \frac{\text{power out (B)}}{\text{power in (A)}} = \frac{0.5 \times 10^{-3}}{1 \times 10^{-3}} = 0.5$$

Half the power that the 1004 Hz test tone introduced at A is lost by the time it reaches B.

Let's repeat the test, this time using less test tone power. The oscillator at demarc A is set to generate 1004 Hz tone at a power of 0.1 mW. At demarc B, the power measurement is 0.05 mW.

The *absolute* power loss is, then,  $0.1 \text{ mW} - 0.05 \text{ mW} = 0.05 \text{ mW}$ .

The relative loss, or the ratio between power out (B) and power in (A), is:

$$\text{Relative Loss} = \frac{\text{power out (B)}}{\text{power in (A)}} = \frac{0.05 \times 10^{-3}}{0.1 \times 10^{-3}} = 0.5$$

The relative loss, or power ratio between B and A, is the same, whether we use a test signal of 1 mW or 0.1 mW.

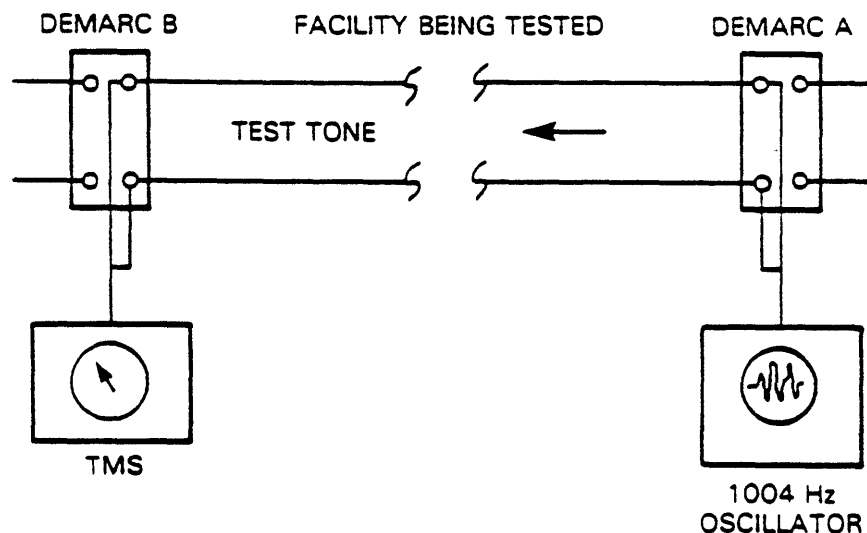


Figure 2-1. Transmission Testing Setup

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This illustrates an important and useful characteristic of transmission circuits:

For a given circuit segment, losses may be expressed in terms of ratios, and the ratio will be the same regardless of the power used over a wide range.

We could, then, express our test results by saying that the power loss between A and B is 1/2.

Although the loss performance of circuit segments *could* be expressed in terms of milliwatts or power ratios, there is a more convenient and useful method of expressing power gains and losses.

## THE DECIBEL

Mathematically, the decibel is a *logarithmic measure*. The logarithm, or log, of a particular number is the mathematical power to which a *base number* must be raised in order to result in the particular number. The *base number* used when we are dealing with the decibel is 10.

For example, what is the logarithm (log) of 100? Another way of asking this question is: To what power do we raise 10 to get 100? The answer is 2 because  $10 \times 10 = 10^2 = 100$ . So,

$$\log (100) = 2$$

Similarly,

$$\log (1000) = 3$$

$$\log (10,000) = 4, \text{ etc.}$$

Logarithms may also be used to express fractional quantities. For example, what is the logarithm of 0.001? Another way of asking this question is: To what power do we raise 1/10 (0.1) to get 0.001? The answer is 3, and, by convention, the log of a fractional number is expressed as negative, so

$$\log (0.001) = -3$$

Logarithms of numbers that are not integral powers of 10 may be calculated by looking them up in a table or by using a hand calculator.

The *decibel* uses logarithms to express power ratios. The decibel is defined as follows:

$$\text{dB} = 10 \log \left( \frac{P_2}{P_1} \right)$$

$P_2$  and  $P_1$  are power measurements expressed in consistent units. The number of decibels is positive if  $P_2$  is greater than  $P_1$ ; the number is negative if  $P_1$  is greater than  $P_2$ . Table 2-1 gives selected power ratios and their corresponding decibel values.

**Table 2-1. Selected Decibel Values**

Power Ratio	dB Value
2	3*
4	6*
8	9*
10	10
100	20
1,000	30
100,000	50
1,000,000,000	90

\*Approximate dB value

## Part II Transmission

In the example illustrated in Figure 2-1, the power ratio between the power measured at B and the power measured at A was one-half. Expressed in decibels:

$$\begin{aligned}\text{Loss, A to B} &= 10 \log (0.5) \\ &= -3 \text{ dB}\end{aligned}$$

By using decibels, we may express the loss or gain of a circuit or piece of equipment without having to explicitly state the actual values of the input and output power. In the example, the loss between A and B will *always* be 3 dB, regardless of the absolute amount of power being transmitted.

### DECIBEL MEASUREMENT RELATIVE TO ONE MILLIWATT

Absolute power is expressed in milliwatts; relative power is expressed in decibels. By establishing a relationship between the decibel and the milliwatt, we can eliminate the milliwatt as an operational unit of measure and deal exclusively with the decibel and related units of measure.

The unit of measure that is used to express absolute power in terms of decibels is *dBm*.

$$\text{dBm} = 10 \log \left( \frac{\text{Power, measured in mW}}{1 \text{ mW}} \right)$$

Since a milliwatt is the standard power reference in communications, it is logical that 0 dBm (the absolute power reference when decibel units are used) is equal to 1 mW of power

Mathematically,

$$\begin{aligned}0 \text{ dBm} &= 10 \log \left( \frac{\text{power out}}{\text{power in}} \right) \\ &= 10 \log (1/1) \\ &= 10 \times 0 = 0\end{aligned}$$

Because the power is an alternating current waveform and impedance can vary as a function of frequency, it is necessary to state what frequency the 0 dBm standard is based upon. The standard frequency is 1004 Hz.

We must also know the resistance or impedance (load) of the circuit. The standard impedance is 600Ω.

Therefore, our reference of 0 dBm is equal to 1 mW of power imposed upon an impedance of 600Ω or a frequency of 1004 Hz.

Tests are usually performed using test signals that are less powerful than 1 mW (0 dBm). Returning once again to the example shown in Figure 2-1, if we apply a 1004 Hz test tone of -13 dBm at A, we will read -16 dBm on the TMS at B. The loss is still -3 dB.

### TRANSMISSION LEVEL POINT

In discussing the performance of a circuit, it is necessary to describe the power at a particular point in a circuit with reference to the power present at other points in the circuit. This power could be signal power, noise or test tones.

Describing this power is similar to describing the height of a mountain (or the depth of the ocean). To measure the height of a mountain, it is necessary to pick a reference height from which to measure. The standard reference height is sea level, which is arbitrarily assigned a height of zero. By measuring all mountains from sea level, comparisons of their height can be made even though they may be many miles apart.



In a similar manner, power at specified points in a circuit may be described in terms of the power at a standard reference point.

This point, which is analogous to sea level, is called the *zero transmission level point*, or 0 TLP.

Any other TLP may be referenced to the 0 TLP by algebraically summing the 1004 Hz gains and losses from the 0 TLP to the point of measurement.

Figure 2-2 illustrates the relationship between TLPs and test tone power in dBm for a simple circuit. The power present at a particular point in a circuit depends on the power at the signal source, on where the source is applied, and on the loss or gain between the two points in question.

Using the 0 TLP concept, the power in a circuit is described by stating what the power would be if it were accurately measured at the 0 TLP. The standard notation here is *dBm0*, which means *power referenced to the 0 TLP*.

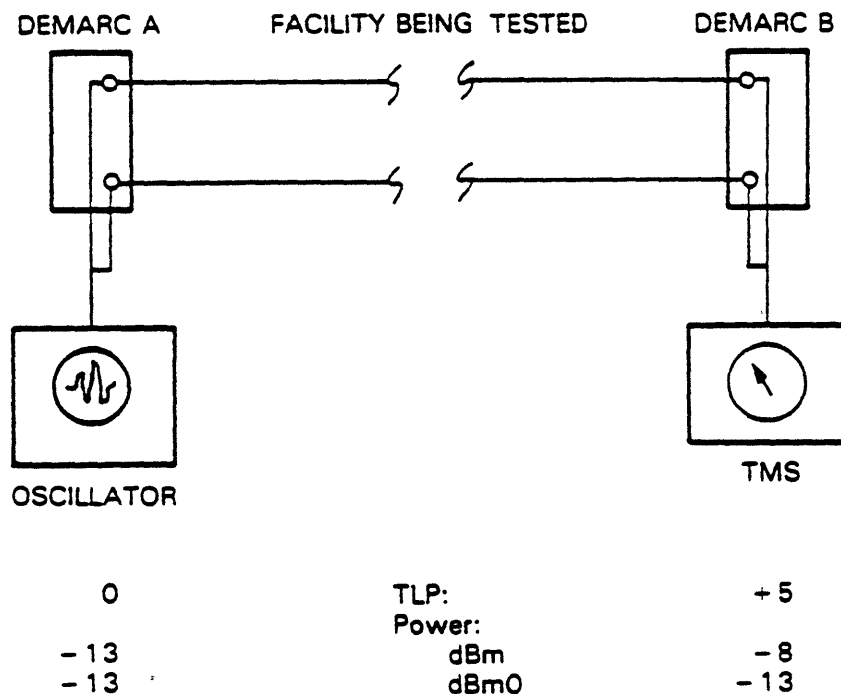
For example, the term  $-13 \text{ dBm0}$  means that the power at the 0 TLP is  $-13 \text{ dBm}$ ; a TMS that is properly set up would measure  $-13 \text{ dBm}$  at the 0 TLP. An example of a  $-13 \text{ dBm0}$  signal is shown in Figure 2-2.

Once the power at the 0 TLP is found, the power at any other point in the circuit can easily be determined. For example, if the signal is  $-13 \text{ dBm}$  when measured at the 0 TLP, it will be 13 dB below the numeric value of any TLP on the circuit when measured at that TLP.

In Figure 2-2, if the signal is  $-13 \text{ dBm}$  at the 0 TLP (making it a  $-13 \text{ dBm0}$  signal), then the power at the  $+5 \text{ TLP}$  can be calculated as follows:

$$\begin{aligned} (\text{TLP}) + (\text{power at the 0 TLP}) &= (\text{power at the } +5 \text{ TLP}) \\ (+5) + (-13 \text{ dBm0}) &= -8 \text{ dBm} \end{aligned}$$

If the  $-13 \text{ dBm0}$  signal is properly measured at the  $+5 \text{ TLP}$ , the meter would read  $-8 \text{ dBm}$ .



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**Figure 2-2. Transmission Level at the 0 TLP**

## Part II Transmission

In a similar manner, if a  $-13$  dBm0 signal were measured at the  $-3$  TLP, the meter would read  $-16$  dBm:

$$\begin{aligned} (\text{TLP}) + (\text{power at the } 0 \text{ TLP}) &= (\text{power at the } -3 \text{ TLP}) \\ (-3) + (-13 \text{ dBm0}) &= -16 \text{ dBm} \end{aligned}$$

To determine the expected power at any given TLP, it is sufficient to know the power present at some other TLP in the circuit. And, just as the mountain does not have to be near the sea in order to determine its height, the 0 TLP does not have to actually exist on the circuit.

Figure 2-3 illustrates a circuit between two demarcs. A  $-29$  dBm test tone signal is applied at the  $-16$  TLP. What should we expect to measure at the  $+7$  TLP?

Even though the 0 TLP does not exist on the circuit, we can describe the power we would see at the 0 TLP if it did exist:

$$\begin{aligned} (\text{TLP}) + (\text{power at the } 0 \text{ TLP}) &= (\text{power at the } -16 \text{ TLP}) \\ (-16) + (\text{power at the } 0 \text{ TLP}) &= -29 \text{ dBm} \\ \text{power at } 0 \text{ TLP} &= -13 \text{ dBm} \end{aligned}$$

Using the relationship again, we can determine the power at the  $+7$  TLP.

$$\begin{aligned} (\text{TLP}) + (\text{power at the } 0 \text{ TLP}) &= (\text{power at the } +7 \text{ TLP}) \\ (+7) + (-13 \text{ dBm0}) &= -6 \text{ dBm} \end{aligned}$$

Use of the 0 TLP reference permits transmission objectives and measured results to be stated independently of any specific TLP, and without specifying what the test tone levels are to be or where the test tone is to be applied.

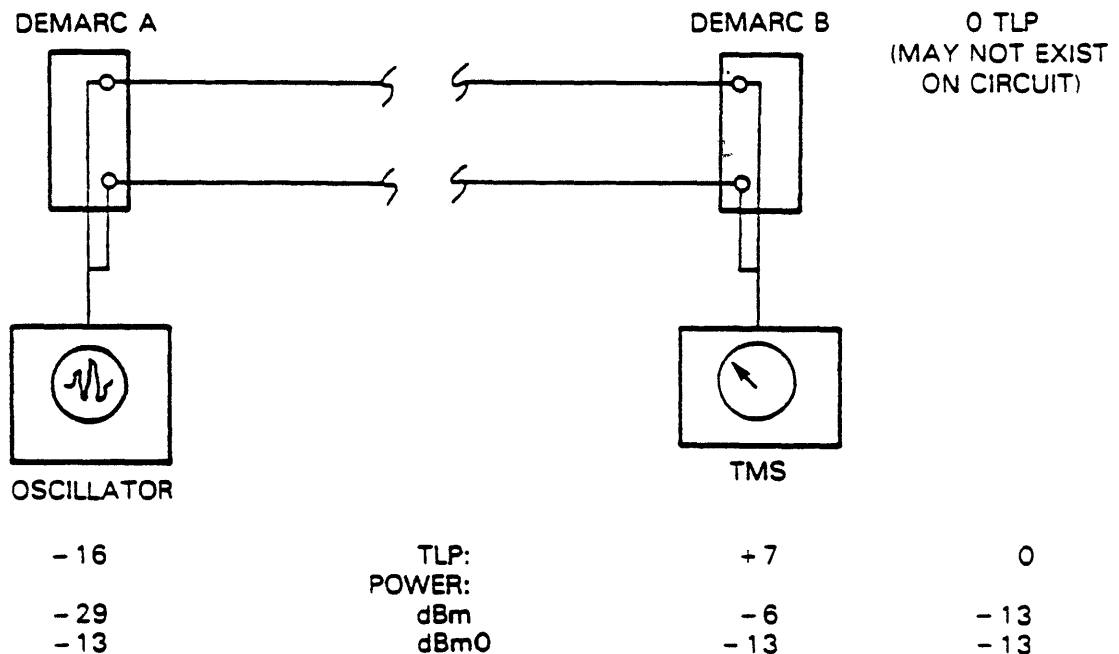


Figure 2-3. Transmission Level at the  $-16$  TLP

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## NOISE MEASUREMENT UNITS

In addition to describing test tone power at various points in a circuit, decibel-related units of measure may be used to describe noise present in a circuit.

**dBm:** To describe power in a circuit, the term dBm is used, meaning "power referenced to 1 mW." Since noise typically contains much less than 1 mW of power, it is convenient to use a reference power that is much smaller than 1 mW. The reference power used in describing noise is  $-90$  dBm.

The notation used to describe noise in terms of reference noise is *dBn*. If we know the noise level in dBm, we can easily measure the noise in dBn:

$$\text{dBn} = \text{dBm} + 90 \text{ dB}$$

For example, a noise measurement of 30 dBn would indicate a power level of  $-60$  dBm (30 dB above the  $-90$  dBm reference noise level). Table 2-2 shows the relationship between dBm0 and dBn.

**Table 2-2. Relationship Between dBm0 and dBn**

dBm0	dBn
0	90
-10	80
-20	70
-30	60
-40	50
-50	40
-60	30
-70	20
-80	10
-90	0

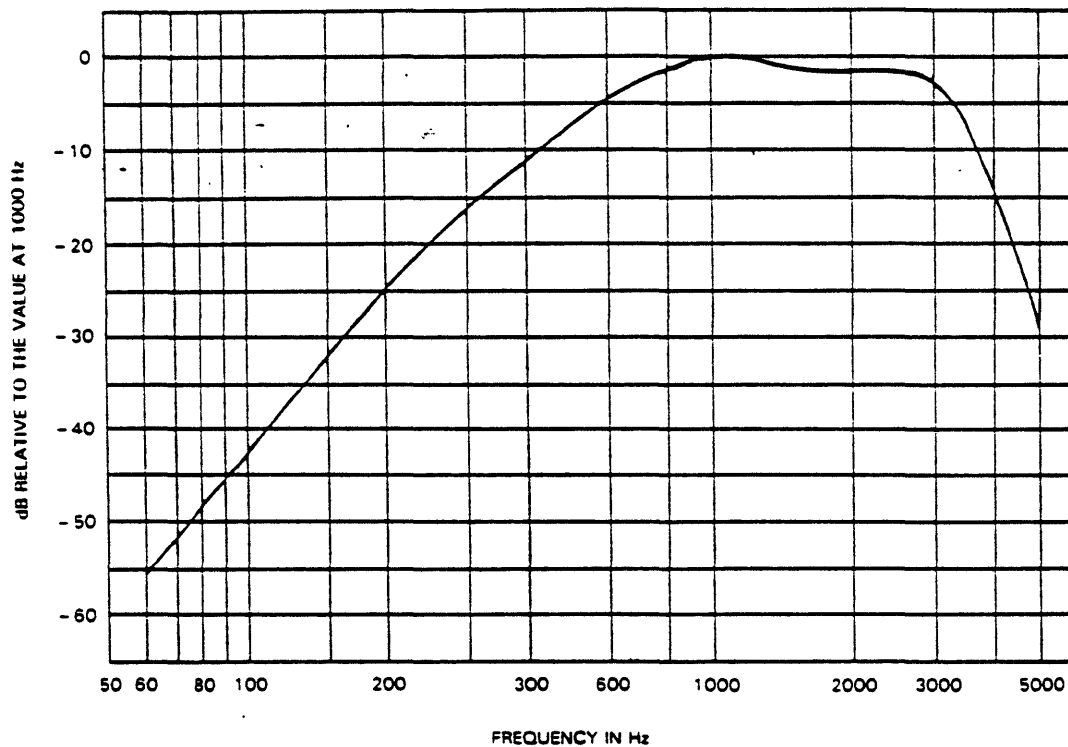
**dBnC:** Noise contains numerous irregular waveforms that have a wide range of frequencies and powers. Although any noise superimposed upon a conversation has an interfering effect, experiments have shown that the interfering effect is greatest in the mid-range of the voice frequency band.

To obtain a useful measure of the interfering effect of noise, the various frequencies contributing to the overall noise are weighted according to their relative interfering effect. This weighting is accomplished through the use of weighting networks, or filters, within Test Measuring Sets (TMS).

Several different weighting schemes are in use; however, we will be primarily concerned with *C-message weighting*. The frequency response curve for C-message weighting networks is shown in Figure 2-4.

Noise measurements through a C-message weighting network are expressed in units of dBnC (noise above reference noise, C-message weighting).

## Part II Transmission



10813A126

Figure 2-4. C-Message Weighting

**dBrnC0:** As with test tone power, noise power may be referenced to the 0 TLP.

For example, if the noise objective for the circuit illustrated in Figure 2-3 is 31 dBrnC0, what is the noise measurement at the +7 TLP?

$$\begin{aligned} (\text{TLP}) + (\text{noise at the 0 TLP}) &= (\text{noise at TLP}) \\ (+7) + (31 \text{ dBrnC0}) &= 38 \text{ dBrnC} \end{aligned}$$

The noise measurement at the +7 TLP is 38 dBrnC.

What is the noise measurement at the -16 TLP?

$$\begin{aligned} (\text{TLP}) + (\text{noise at the 0 TLP}) &= (\text{noise at TLP}) \\ (-16) + (31 \text{ dBrnC0}) &= 15 \text{ dBrnC} \end{aligned}$$

The noise measurement at the -16 TLP is 15 dBrnC.

## **CHAPTER 2 REVIEW QUESTIONS**

1. Why is the milliwatt used as the basic power unit in transmission, rather than a larger unit?
2. Why is the decibel used in transmission measurements?
3. Is the loss (or gain) in a circuit, when expressed in dBs, dependent upon the absolute power carried by the circuit?
4. What does a power difference of  $-3$  dB mean?
5. What power gain does 10 dB represent?
6. Define dBm.
7. If a 0 dBm 1004 Hz test tone is applied at the 0 TLP, what would the power measurement be at the +7 TLP?
8. If a power measurement of  $-6$  dBm is taken at the +7 TLP, what would the absolute power measurement be at the 0 TLP?
9. What does a noise measurement of 58 dBm represent in dBm?
10. If the end-to-end noise objective for a circuit is 35 dBmC0, what is the maximum allowable noise at the +7 TLP?
11. Define the 0 dBm standard.



## CHAPTER 3

# TRANSMISSION PERFORMANCE PARAMETERS

### INTRODUCTION

This chapter contains a general discussion of parameters that affect transmission quality. The parameters that will be covered may be categorized as follows:

- Loss
- Signal Distortion
  - Attenuation Distortion
  - Frequency Shift Distortion
  - Delay Distortion
- Noise
  - Idle Channel Noise
  - Signal-dependent Noise
  - Impulse Noise
  - Crosstalk
- Echo

Transmission impairments always exist. The essence of good design and standards is to reduce impairments to acceptable levels. Test results indicating that any of these parameters fall outside of limits specified by the circuit design may be evidence of unacceptable transmission impairments.

The influence that each of the parameters has upon transmission will be discussed. It is well beyond the scope of this text to discuss the details of the operation of transmission line equipment; however, devices and techniques that may be used to control transmission impairments are briefly described.

Although the emphasis here is on the influence of transmission parameters upon speech transmission, the effects of the parameters upon data transmission are mentioned where appropriate.

### LOSS

In a telephone connection, the volume of the received sound must be such that the intelligibility and naturalness are satisfactory. If the volume is too low, some sounds will be misunderstood or not heard. If the volume is too high, the overloading of the ear will cause distortion within the ear, reducing intelligibility.

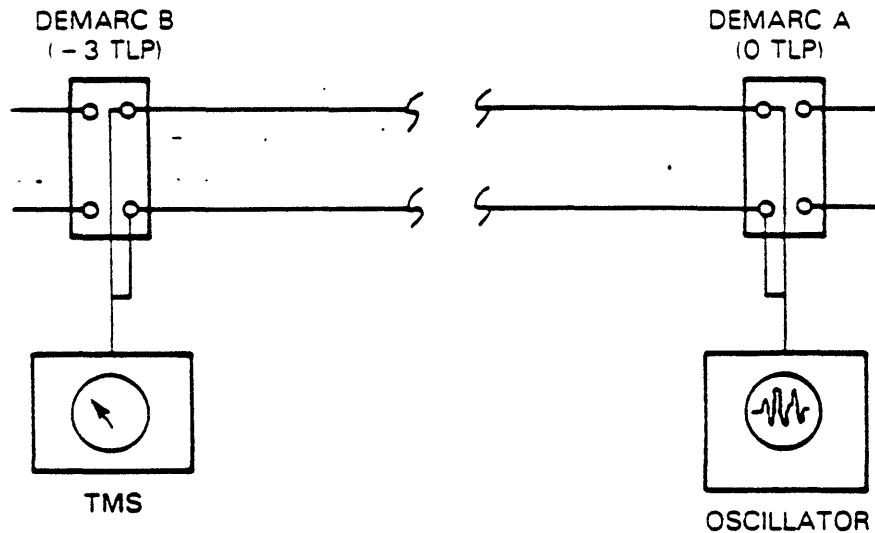
A certain amount of loss is designed into each voice circuit to minimize the effects of certain impairments while achieving acceptable volume.

The transmission parameter that has the greatest influence on listener volume is *transmission loss*. Loss is the dissipation of energy that naturally occurs due to the electrical characteristics of transmission facilities and equipment.

The testing setup for measuring the loss of a circuit segment is illustrated in Figure 3-1

A 1004 Hz test tone is applied at a test access point at one end, and the receive level is measured at the other end.

## Part II Transmission



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Figure 3-1. Loss Measurement

The difference in power, expressed in decibels, is the *Actual Measured Loss* (AML) between the two ends of the circuit under test. The *Expected Measured Loss* (EML) is the design loss of the circuit. The difference between the design loss and the AML is the *AML deviation*. The AML deviation is very important, as it indicates how well the loss characteristics of the circuit conform to design specifications.

In Figure 3-1, the test tone is applied at the 0 TLP and measured at the -3 TLP. The EML is 3 dB.

Example:

If a 0 dBm test signal is applied at point A and -3.2 dBm is observed at B, what is the AML deviation?

$$\begin{aligned}
 \text{EML} &= -3.0 \text{ dBm} \\
 \text{AML} &= -(-3.2) \text{ dBm} \\
 \text{AML Deviation} &= \frac{\quad}{0.2 \text{ dB (more loss)}}
 \end{aligned}$$

When the AML deviation is *positive*, the loss is greater than the circuit design specifies.

The allowable limits for AML deviation should be specified by the person who designed the circuit that is being tested.

To meet end-to-end circuit design objectives, it is sometimes necessary to decrease or increase the loss in a particular portion of a connection.

Loss may be increased by using *resistance networks* (usually referred to as *pads*). The configuration of the resistors determines the type of pad. The most common types employed in telephone equipment are the T, H, Pi and square pads. The amount of loss that a pad introduces is given in decibels. (The pad designation refers to the configuration of the resistance network.)

It is often necessary to provide amplification to compensate for losses incurred in the transmission medium. The electronic device used for this purpose may be compared to the audio amplifier of a radio receiver or a record player. These amplifiers and their associated circuitry are called *repeaters*.



## SIGNAL DISTORTION

Speech (and data) transmission is composed of a number of frequency components that add together to form a complex waveform. The various frequency components that make up the complex waveform bear a very definite relationship to one another, both in magnitude and in time. Alteration of the relationships between the various waveforms distorts the composite waveform, and, if the distortion is excessive, transmission impairments may result.

**ATTENUATION DISTORTION:** One requirement for a channel is to provide uniform transmission loss characteristics for all of the frequencies making up the complex waveform. Typical channels have some variation in loss with respect to frequency.

*Attenuation distortion*, also referred to as *slope*, is the difference in loss at one frequency with respect to the loss at another frequency. In speech transmission, excessive attenuation of the lower frequencies will cause the reproduced voice to sound "tinny." Attenuation of the higher frequency will lower the perceived pitch and will reduce intelligibility.

A simple way to measure slope is to measure the loss at 404 Hz and 2804 Hz and compare it to the loss at 1004 Hz. The larger the difference the greater the slope. The actual measured loss for all three frequencies is used in these calculations.

Example:

The following loss measurements were taken. What is the slope?

404 Hz	- 3.0 dBm
1004 Hz	- 3.1 dBm
2804 Hz	- 4.0 dBm

Subtract the 404 Hz loss from the 1004 Hz loss:

$$\begin{array}{r}
 \text{AML, 1004 Hz} = -3.1 \text{ dBm} \\
 \text{AML, 404 Hz} = -(-3.0 \text{ dBm}) \\
 \hline
 -0.1 \text{ dB (less loss)}
 \end{array}$$

Then subtract the 1004 Hz loss from the 2804 Hz loss:

$$\begin{array}{r}
 \text{AML, 2804 Hz} = -4.0 \text{ dBm} \\
 \text{AML, 1004 Hz} = -(-3.1 \text{ dBm}) \\
 \hline
 0.9 \text{ dB (more loss)}
 \end{array}$$

The slope is 0.9 dB at 2804 Hz. It is positive because there is more loss at 2804 Hz than at 1004 Hz.

The allowable limits for attenuation distortion are specified by the circuit design.

Where cable is used, excessive attenuation distortion at higher frequencies may be reduced by increasing the distributed inductance of the cable. This is accomplished by placing inductors of a specific size (specified in *millihenrys*, or *mH*) at specific intervals along the cable. The inductors are called *load pots*, or coils, and when they are used the cable is said to be *loaded*.

A common loading procedure is to place 88 mH coils at 6000-foot intervals. This is called H88 loading, where

H designates the 6000-foot spacing

and

88 refers to the size of the inductance.

## Part II Transmission

Usually the gauge of the cable is also stated; i.e., 22 H88 means 22 gauge cable loaded as above.

While the design of cable facilities is beyond the scope of this text, it is necessary to understand the terms in order to perform certain tests and to make certain equipment adjustments.

Attenuation distortion can be compensated for by the use of *equalizers*. An equalizer is a filter that has a frequency response that removes the attenuation distortion inherent in the transmission facility.

**FREQUENCY SHIFT DISTORTION:** When part of an end-to-end connection transverses carrier facilities, the frequency of the signal may be shifted either upward or downward. This is called *frequency shift distortion*.

Frequency shift distortion usually has minimal effect on speech transmission, as the human ear is not very sensitive to small changes in frequency. However, it can be a serious problem in data transmission if the shift is large.

To measure the frequency shift parameters, a frequency counter is used at one end of a circuit to measure a known frequency sent from the other end.

**DELAY DISTORTION:** Waves of different frequencies travel at different velocities over transmission facilities, particularly metallic facilities. As a result, some of the waves making up the composite waveform arrive at the destination before others, distorting the composite waveform. Another way of stating this is that phase relationship between the component waveforms is different at the receive point than at the transmit point. This is called *delay distortion* or *phase distortion*.

Phase distortion usually has little effect upon voice transmission because the human ear is insensitive to small changes in phase. Data transmission may be seriously affected due to the alternation of the component waveforms that form the digital pulse.

Delay distortion itself is difficult to measure. An approximate measure of delay distortion is the envelope delay test, which measures the relative delay among the frequencies in the voice frequency band. Envelope delay measurements are expressed in microseconds ( $\mu\text{s} = 0.000001 \text{ sec}$ ). The envelope delay measurement represents the difference in arrival times between the slowest moving wave (or frequency) and the fastest moving wave.

The allowable limits for envelope delay will be specified by the circuit design.

Delay distortion, or phase distortion, can be compensated for by using *delay equalizers*. Equalizers impose delay on the faster frequencies, restoring the phase relationships existing at the transmit point.

## NOISE

*Noise* may be generally defined as introduced energy that interferes with or masks desired transmission.

In transmission systems there are two types of noise sources. The first is noise that is inherent in the transmission system and is independent of the signal being carried by the system. This type of noise is called *idle channel noise* and has a number of different sources. The second type of noise is *signal-dependent noise*, which is generated by circuit components as a result of an actual signal being transmitted.

Any transmitted signal represents a certain degree of order or pattern in the electrical waveform. The energy contained in noise generally has no particular pattern; however, in the case of crosstalk, it may have an unwanted, distracting pattern. At low but discernible levels noise is annoying. As noise becomes excessive, the desired signal will eventually become "buried" in the noise, thus destroying communication.

There are many sources of noise, each having its own effect upon speech and data transmission. Four categories of noise are discussed here: idle channel noise, signal-dependent noise, impulse noise, and crosstalk.

**IDLE CHANNEL NOISE:** Idle channel noise is a catchall category that includes all sources that produce noise in a fairly constant manner in the absence of a signal. Other terms for idle channel noise are background noise, random noise, and message circuit noise.

A major component of idle channel noise is *thermal noise*. Noise is generated when electrons collide with some of the molecules of the conductor; as the temperature of the conductor is increased, noise also increases as more of the electrons collide with the more agitated molecules of the conductor.

Another fundamental source of idle channel noise is *electromagnetic coupling*. Almost all transmission circuits are exposed to external electrical influences due to their proximity to other circuits. Loops and trunks are usually physically close to other circuits in cables or lines. Any circuit passing through a Central Office is potentially exposed to many other circuits and may also be affected by battery noise. Many circuits have power transmission lines paralleling part of their routes. Exposure to electromagnetic fields created by the currents in these nearby, *disturbing circuits* may cause *induced currents* in the *disturbed circuit*.

When these induced currents are small, their source may not be readily identifiable, and their resulting noise will contribute to idle channel noise. (Some noise caused by coupling is readily identifiable, such as crosstalk.)

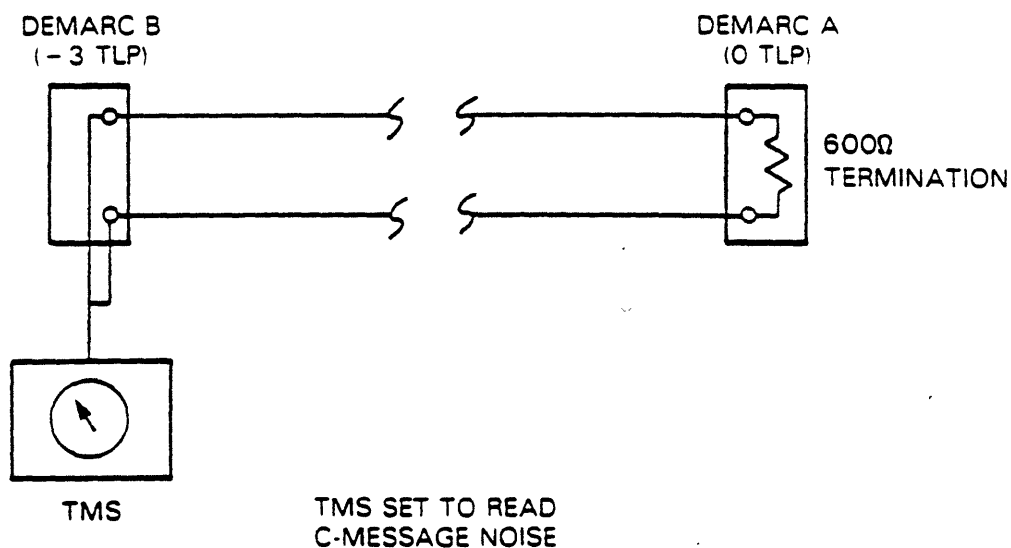
Idle channel noise may affect both speech and data transmission. However, it is usually more critical in speech transmission. This is because speech has a wider range of power levels than does data transmission, and the human ear is more sensitive to low-level noise than are data-sets.

In Chapter 2, the logarithmic units used to measure noise, dBrn, dBrnC and dBrnC0, were discussed. The allowable level of idle channel noise will be specified by the circuit design. This ensures that the normal speech sounds will be acceptably louder than the idle channel noise.

The test for idle channel noise is called the C-message noise test. (C-message is an abbreviation for Circuit-message.) As mentioned in Chapter 2, the noise measurement set used for the test contains a C-message filter. This filter eliminates noise components that have little effect on the ear, while passing the frequencies that have the greatest impairing effect on speech transmission.

Example:

Figure 3-2 illustrates the general setup for measuring C-message (idle channel) noise. At point A (the 0 TLP) the circuit is terminated with a 600 resistor. At point B (- 3 TLP), a noise measuring set is attached to the circuit. A reading of 30 dBrnC is taken. To adjust this reading to dBrnC0, 3 dB is added to the reading, giving 33 dBrnC0.



**Figure 3-2. Idle Channel Noise Measurement Setup**

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## Part II Transmission

**SIGNAL-DEPENDENT NOISE:** Background noise exists regardless of the presence or absence of any signal (thus the term "idle channel noise"). When a signal is present on a circuit, the signal itself will generate a certain amount of noise, particularly when transmission equipment such as repeaters and equalizers are present.

The effects of signal-dependent noise on both speech and data transmission are similar to the effects of background noise.

The test used to measure signal-dependent noise is called the *C-notched noise* test or the *signal-to-noise (S/N)* test. Two measurements are required for this test.

The first measurement is similar to the C-message noise test (covered previously) except that a  $-13$  dBm0 1004 Hz test tone is applied at point A, as shown in Figure 3-3 (The results of the S/N test will vary depending upon the test tone power used (in dBm0). Signal-to-noise design limits are often based upon a test signal of  $-13$  dBm0. Therefore, the discussion here will assume a test signal of  $-13$  dBm0.) At point B, the noise is recorded in dBnC.

For the second measurement a different filter, called a *C-notched filter*, is used at B. The C-notched filter has the same characteristics as the C-message filter, except that it also employs a very narrow band-elimination filter that removes the test tone. This measurement is also recorded in dBnC.

It is not necessary to reference either measurement to the 0 TLP (i.e., the dBnC measurements do not need to be converted to dBnC0). The difference between the first and second measurements is the "test tone to C-notched noise ratio," and is an *approximation* to the signal-to-noise ratio (The results of this test are often called the S/N ratio.)

The allowable limits for this test are specified by the circuit design.

Example:

Figure 3-3 illustrates the setup for measuring C-notched noise.

Point A is the  $-3$  TLP, so a 1004 Hz test tone at  $-16$  dBm ( $-13$  dBm0) is applied at A. At B, a reading of 74 dBnC is obtained using a C-message filter, and a reading of 50 dBnC is obtained with a C-notched filter. The S/N ratio is  $74 - 50 = 24$  dB.

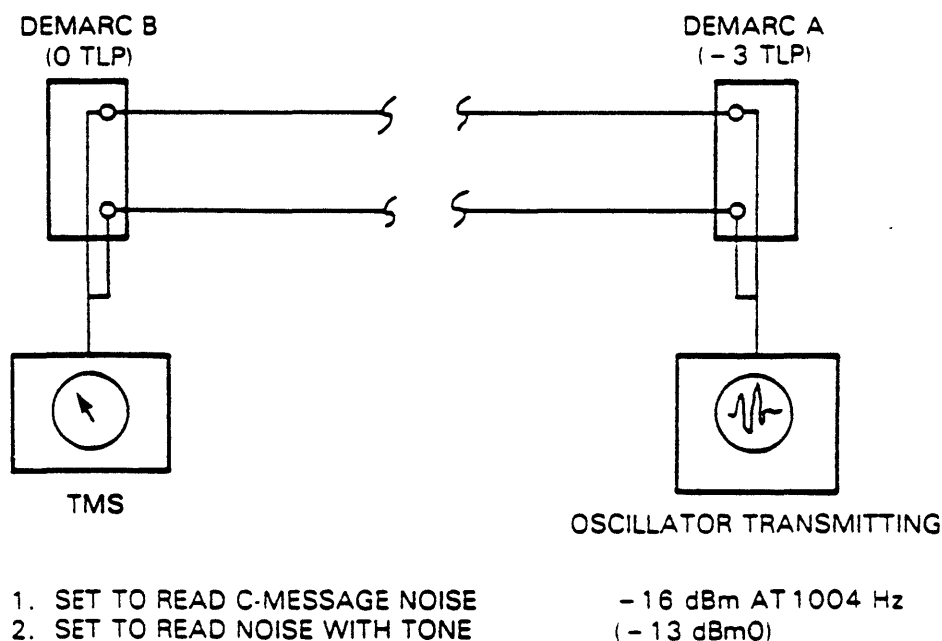


Figure 3-3. C-Notched Noise Measurement

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**IMPULSE NOISE:** Impulse noise is characterized by large peaks, or impulses, in the total waveform. Impulse noise has many sources, some of the more important being lightning, relay and switch operation, microwave path fading on carrier channels, and "crackling" due to thermal effects in cable.

Impulse noise usually has little effect on speech transmission, as the human ear can tolerate it. Data transmission, however, is very sensitive to impulse noise. Depending on the impulse amplitude, duration and time of occurrence, individual data bits or blocks of data may be obliterated, resulting in errors in the received signal.

**CROSSTALK:** Whenever telephone circuits follow adjacent paths, they are susceptible to crosstalk. Crosstalk may be produced by inductive or capacitive coupling in parallel lines or at junctions, producing unwanted signals in the disturbed circuit.

Crosstalk varies greatly in amplitude and intelligibility. Low-level crosstalk may be perceived as noise. Intelligible crosstalk is particularly objectionable, as it reduces the security of the telephone system.

## ECHO

Echo results when the talker's energy is reflected back from some distant point in the transmission channel. Echoes constitute one of the most serious forms of impairment in telephone circuits, whether the circuits are used for speech or data transmission.

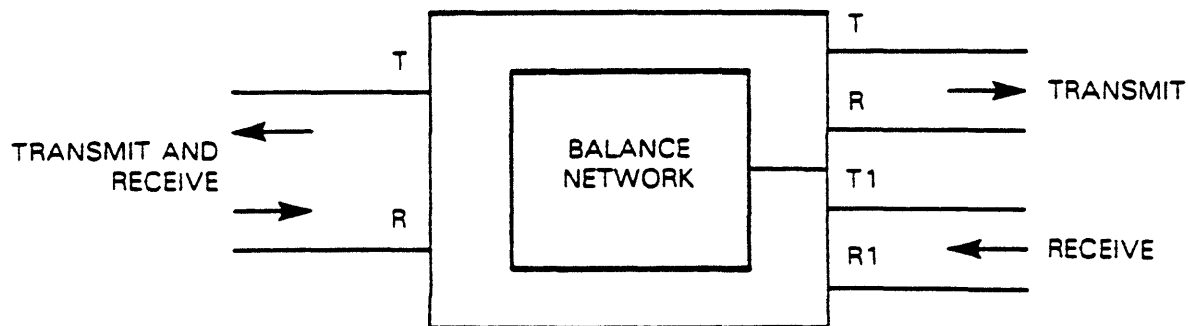
The most common source of echo is the interface between 2-wire and 4-wire facilities

As previously discussed in Part I, 2-wire circuits consist of two conductors called tip and ring. These circuits are capable of transmitting and/or receiving speech or data signals from either end of the circuit. While economical, 2-wire circuits have characteristics that allow for limited-distance use only.

When this range is exceeded, or when for other design reasons it becomes necessary, 4-wire circuits must be used. These circuits use two wires for each direction of transmission. (See Figure 3-4.) The two conductors called tip and ring are used in the transmit direction; the two conductors called tip 1 and ring 1 are used in the receive direction.

When interconnection of the two facilities is necessary, a *hybrid circuit* (commonly called a 4-wire termination set, or 4 WTS) is used. The hybrid circuit uses four signal paths for connection and impedance matching

- RECEIVE PATH—receives signals from the 4-wire facility.
- TRANSMIT PATH—transmits signals to the 4-wire facility.
- 2-WIRE PATH—connects the 4 WTS to the 2-wire facility
- BALANCE NETWORK—matches the 4 WTS impedance to the 2-wire facility's impedance.



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**Figure 3-4. 4-Wire Termination Set**

## Part II Transmission

Two forms of echo impairment, called *talker echo* and *listener echo*, are frequently encountered at the junction of 4-wire and 2-wire circuits. The most serious form is *talker echo* which can be defined as hearing one's own speech being returned. It is common for a connection to have two 4-wire termination sets, as in Figure 3-5. The other form of echo impairment, *listener echo*, is the result of energy being reflected back toward the talker by the distant-end termination set, and the reflected energy then being reflected again toward the listener by the near-end termination set. When talker echo is satisfactorily controlled, listener echo is seldom a problem.

The significance of the echo impairment is directly proportional to the distance of the overall connection as this relates to the time required for the signal to be reflected.

If the elapsed time is short, the reflected signal arrives faster than the ear can comprehend the intelligence of the signal. The effect is that the listener may perceive an increase in the sidetone volume. For very short distances, the listener might not even be aware of the echo's presence.

At medium distances, the talker may have the sensation of "speaking into a barrel." As the distance increases, the reflected signal arrives later and can be perceived by the talker as his own speech. The talker may think that the distant party is trying to interrupt, and this can interfere with the talker's normal speech process.

Listener echo is the double reflection of the signal and is generally of low-signal amplitude. If listener echo is significant, it can cause signals to be sustained or regenerated by the transmission path to create a condition called singing.

Singing is analogous to the "howl" that can be generated in a public address system when a microphone is placed too close to a speaker.

The overall effect of echo impairments depends upon the following:

- How loud the echo is
- How long the echo is delayed by the transmission path
- The speaker's tolerance to the echo phenomenon

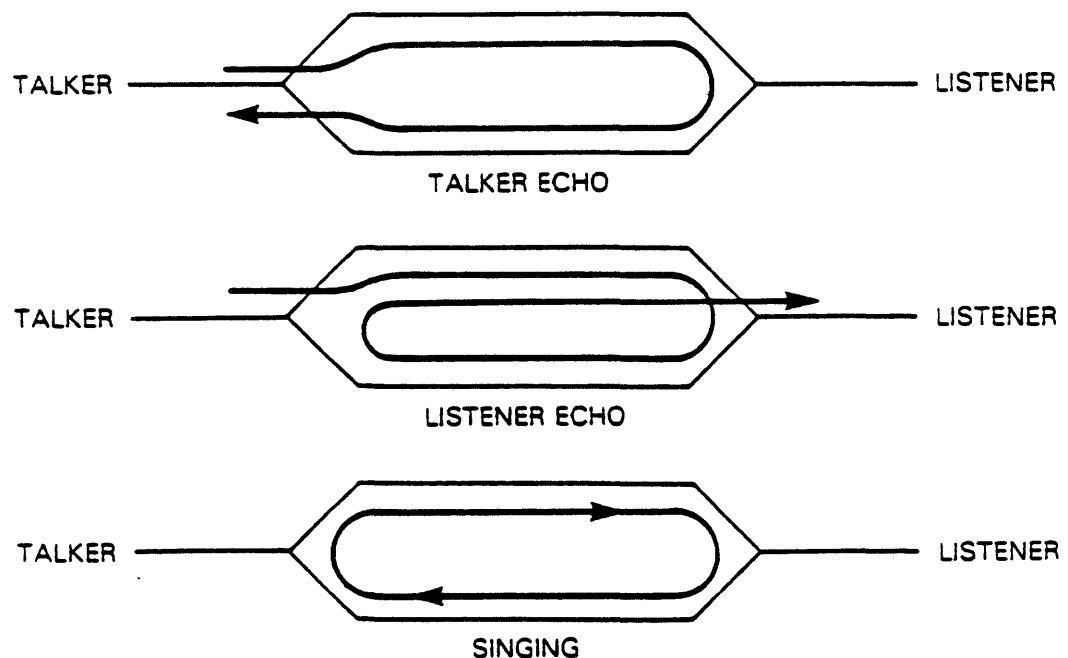


Figure 3-5. Echo Pathways

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**BALANCE THEORY:** A diagram of a typical hybrid arrangement is shown in Figure 3-6. Two of the impedances in the 4-wire circuit,  $Z_T$  and  $Z_R$ , are usually under design control, and normally there isn't a problem in achieving a good match between them. The match between the balancing network,  $Z_N$ , and the 2-wire facility,  $Z_{2W}$ , may not be under design control and can be difficult to match.

When the hybrid connects to a dedicated circuit on the 2-wire side, impedance  $Z_{2W}$  can be under design control or at least has some known value. When this occurs, the balancing network's impedance,  $Z_N$ , may be adjusted to match  $Z_{2W}$  to some degree.

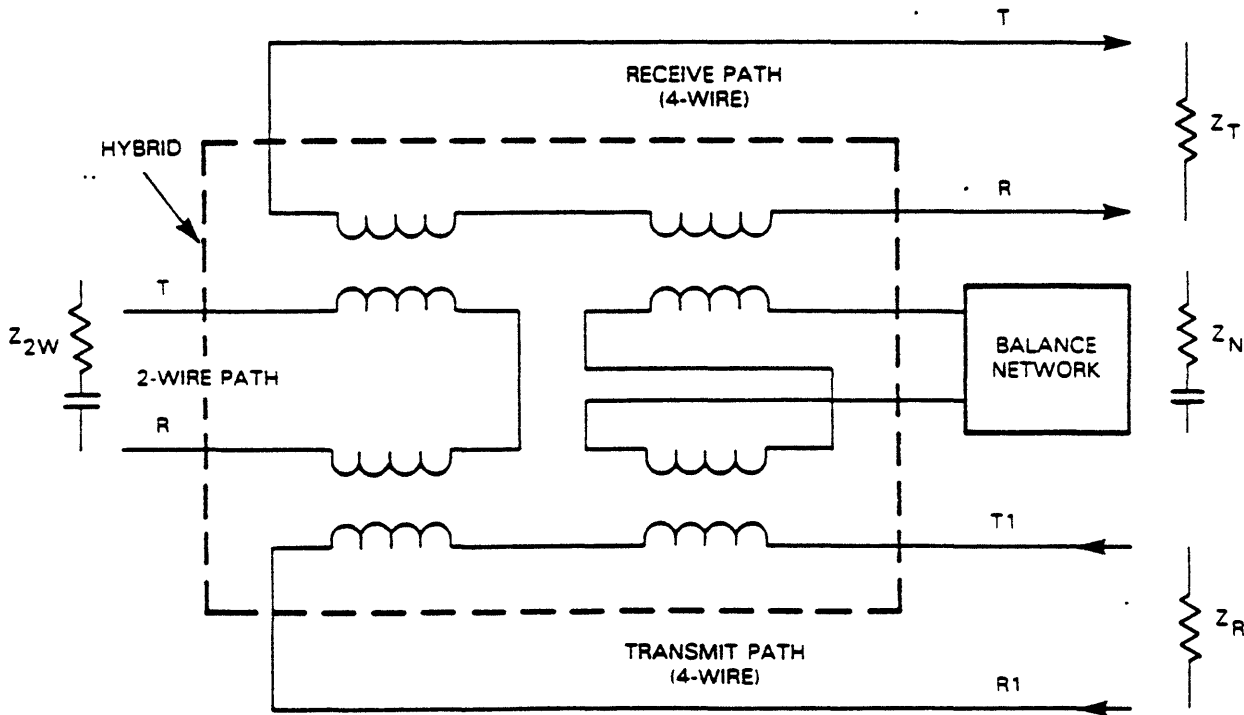
When the hybrid's 2-wire path doesn't connect to a dedicated circuit, impedance  $Z_{2W}$  isn't under design control and will vary, subject to actual switched connections. The balancing network's impedance can only be adjusted for some approximate value. Since this is only a compromise value of the actual impedance, some energy from the receive path will be reflected back into the transmit path.

The amount of reflection is directly proportional to the degree of mismatch between  $Z_N$  and  $Z_{2W}$  and can be expressed in decibels as:

$$\text{Return Loss} = 20 \log \left( \frac{(Z_N + Z_{2W})}{(Z_N - Z_{2W})} \right)$$

The formula, stated here in simplified form, assumes that both  $Z_T$  and  $Z_R$  are equal.

The higher the Return Loss figure, the better the performance. For example, a Return Loss value of 20 dB is better than a value of 0 dB.



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**Figure 3-6. A Typical Hybrid Arrangement**

## Part II Transmission

**DEFINITION OF TERMS:** There are several terms used to test and express the degree of balance. The following definitions of commonly used terms are provided as an aid to understanding these terms:

1. *Trans-Hybrid Loss (THL)*

If the 2-wire side of the hybrid is totally mismatched to the balance network (such as applying a tip/ring short), total reflection of power to the transmit path occurs. This is generally referred to as Trans-Hybrid Loss (THL) and is used to calibrate the balance test and to verify circuit equipment operation.

The actual THL value observed depends on the following:

- The type of hybrid
- Test level point differences between the test instrument and the circuit under test
- The losses/gains of the transmit and receive paths

2. *Echo Return Loss (ERL)*

If sufficient power is reflected back to the talker within the range of 500 to 2500 Hz, the talker may hear his own voice and become confused. This effect is called talker echo and is expressed in terms of Echo Return Loss (ERL), the loss of the reflected signal.

3. *Singing Return Loss (SRL)*

The ability of the hybrid to "sing" or sustain a signal is expressed in terms of Singing Return Loss (SRL). SRL is measured in two bands located between 200 to 500 Hz and 2500 to 3200 Hz using a weighted-noise technique. The bands are called SRL LO and SRL HI, respectively. The worst band (lowest numerical value in decibels) becomes the SRL measurement used in analysis.

4. *Single Frequency Return Loss (SFRL)*

Single Frequency Return Loss (SFRL) measures the ability of the circuit to sing or oscillate at a single frequency in the 200 to 3200 Hz voice band. The SFRL applies to the frequency that has the poorest return loss. It is the critical frequency at which gain and phase relationships are the poorest and may cause singing.

While, in theory, singing can occur at any frequency, it is usually found at the low and high ends of the voice band due to the 2-wire facility's characteristics.

The difference between SFRL and SRL tests is in the method of measurement. The SFRL test finds the critical frequency that can cause singing, while the SRL test uses a weighted-noise technique of the two bands to determine the overall performance.

Either test method is acceptable, but the general practice today calls for making SRL measurements.

**BALANCE TESTING:** To test balance, the following tests are conducted:

- Trans-Hybrid Loss (THL)
- Echo Return Loss (ERL)
- Singing Return Loss High/Low (SRL HI/SRL LO)

If a short is placed across the 2-wire line at the hybrid coil and a known amount of power is sent into the receive leg, the reflected power can be measured at the transmit leg. For example, if a frequency of 1000 Hz at a power of 0.0 dBm is sent into the receive leg and the power measured at the transmit leg is -7.5 dBm, the total 1000 Hz loss from the receive leg input to the transmit leg output is 7.5 dB. This is



the loss of the hybrid from the receive leg to the transmit leg and is generally referred to as the Trans-Hybrid Loss (THL), with 0 dB return loss (total reflection of power) at the 2-wire terminals. The actual measured trans-hybrid loss will depend on the type of hybrid and the loss of the receive and transmit pads, if any.

Once the short has been removed, the 2-wire side of the hybrid has been connected to its 2-wire facility, and the distant end of the 2-wire line facility has been terminated in the correct impedance, another measurement is taken with the same input power and frequency at the receive leg. The power measured at the transmit leg will be lower than with the 2-wire leg shorted, because the network and the 2-wire line facility with its terminating impedance will absorb most of the power sent into the receive leg. For example, if the power measured at the transmit leg is now -26.5 dBm, the loss from the receive leg to the transmit leg is 26.5 dB. Since the losses of the hybrid and pads (if any) are common to both tests, the 7.5 dB loss for the 0 dB return loss condition is subtracted from the 26.5 dB, leaving 19.0 dB. This 19.0 dB is the return loss and represents the degree of balance between the 2-wire facility and the balancing network at the frequency of the applied power. Higher degrees of balance are indicated by larger values of return loss.

$$\begin{array}{rcl} \text{Circuit Terminated Correctly} & = & -26.5 \text{ dBm} \\ \text{Circuit Shorted at Tip \& Ring} & = & -(-7.5) \text{ dBm} \\ \hline \text{Return Loss} & = & 19.0 \text{ dB} \end{array}$$

When balance measurements are made with ERL Test Sets like the Wiltron 9041, the THL measurement is made with the equipment set to read ERL. With the 2-wire facility shorted at the hybrid, the Wiltron meter is adjusted to zero by using the THL adjust switches. The sum of the final switch readings is the THL.

To measure the ERL, the short is removed and the 2-wire facility is terminated with the correct impedance. With the Test Set still in the ERL test mode, the ADD dB switch is adjusted until an on-scale reading is obtained on the meter.

The sum of the ADD dB switch setting and the meter reading is the true ERL, as the THL was automatically subtracted by the Test Set.

The Test Set is then placed in the SRL LO mode and a reading is obtained using the same procedure as the ERL test (do not make another THL test or adjust the THL switches). The Test Set is then placed in the SRL HI mode and readings are taken again. The lowest numerical value read is equal to the SRL.

A frequency sweep of the voice band between 200 and 3200 Hz is made while the test circuit is still terminated in the correct impedance. The frequency exhibiting the poorest loss determines the SFRL. The THL is subtracted from the value read on the Transmission Measuring Set, in order to obtain the SFRL value. Usually, the SFRL test is not necessary if SRL tests have been made.

## **CHAPTER 3 REVIEW QUESTIONS**

1. The following measurements were taken. What is the AML deviation?

$$\text{EML} = -3.5 \text{ dBm}$$

$$\text{AML} = \underline{-3.0 \text{ dBm}}$$

$$\text{AML Deviation} = ?$$

2. The following measurements were taken. What is the slope?

$$404 \text{ Hz} \quad -3.5 \text{ dBm}$$

$$1004 \text{ Hz} \quad -4.0 \text{ dBm}$$

$$2804 \text{ Hz} \quad -4.2 \text{ dBm}$$

3. What is the equipment called that may be used to correct attenuation or envelope delay distortion?
4. How does noise affect voice transmission?
5. The following measurements were taken. What is the S/N ratio?

$$\text{noise with tone} = 60 \text{ dBmC}$$

$$\text{C-notched noise} = 38 \text{ dBmC}$$

6. Draw a sketch that illustrates echo pathways.
7. What does THL measure?
8. How is echo controlled?

## CHAPTER 4 TELCO INTERFACES

### INTRODUCTION

This chapter provides a brief discussion of the conventions used in the interconnection of customer-provided equipment, such as PBXs, with the public telephone network transmission facilities.

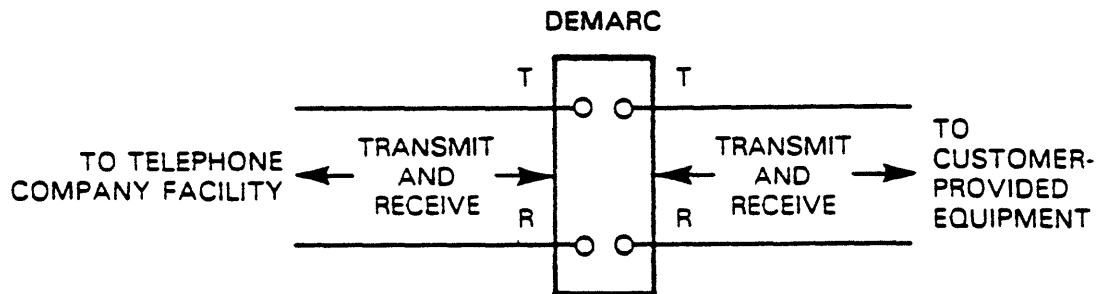
Specific connectors, jacks and plugs are not discussed. Allowable interconnection configurations and their mechanical and electrical specifications are thoroughly described in the Federal Communications Commission Rules and Regulations—Part 68 (Connection of Terminal Equipment to the Telephone Network). The Bell System Technical Reference #47101 describes the standard jack arrangements to be provided by Bell operating companies for the connection of telephone-company and customer-provided registered equipment to the public switched network.

Interfaces provide definite points where equipment owned by the telephone company may be physically separated from customer-owned equipment. When signaling or transmission problems occur, testing in both directions from the interface can be conducted to determine who is responsible for the problem. Use of standard terminology and conventions with reference to interfaces is essential if testing and troubleshooting are to be successful.

In this chapter, the following interfaces are discussed: 2-wire, 2-wire with E&M signaling leads, and 4-wire with E&M signaling leads.

### 2-WIRE INTERFACE

As shown in Figure 4-1, the 2-wire circuit consists of two leads. These leads are called tip and ring and provide a direct-current path for signaling as well as a bidirectional, or *full-duplex*, path for transmission.



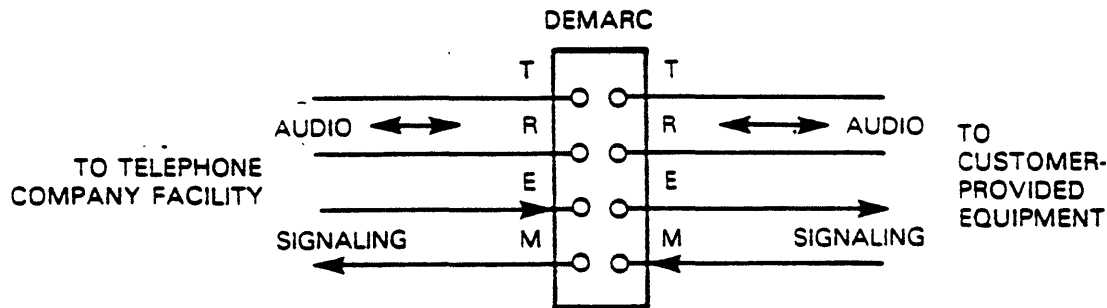
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Figure 4-1. A 2-Wire Interface

### 2-WIRE WITH E&M INTERFACE

As shown in Figure 4-2, the 2-wire with E&M interface actually consists of four wires: tip, ring, E lead, and M lead. The tip and ring leads form the two-way audio path. The E&M leads are used for direct-current signaling. Signaling from the customer equipment toward the demarc is conducted over the M lead; signaling from the demarc toward the customer equipment is over the E lead (E&M lead operation is discussed in Part I of this book.)

**Part II  
Transmission**



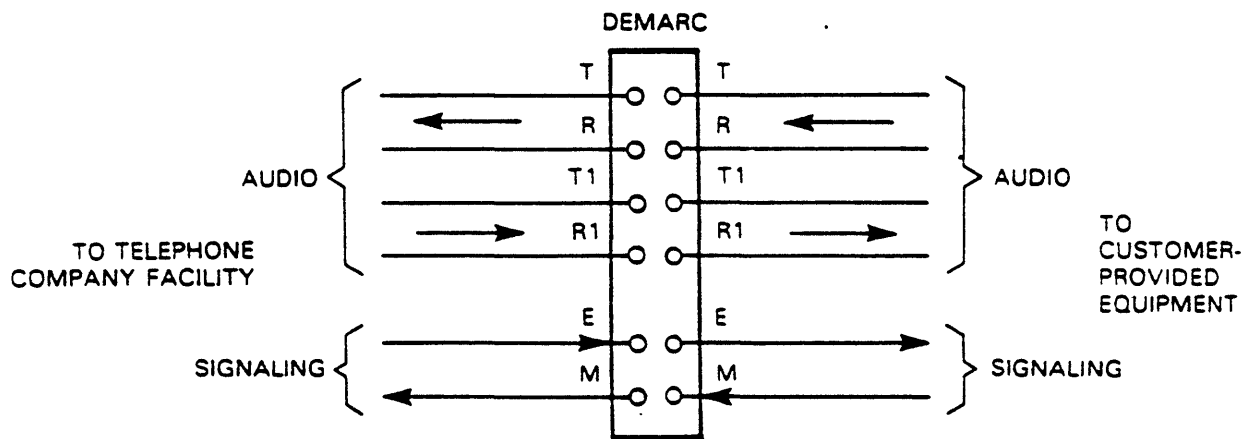
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Figure 4-2. A 2-Wire with E&M Interface

**4-WIRE WITH E&M INTERFACE**

The 4-wire with E&M interface actually consists of six conductor leads (or three pairs), as shown in Figure 4-3. Audio transmission toward the demarc from the customer equipment is carried over the T and R leads; audio transmission from the demarc toward the customer is over the T1 and R1 leads.

Signaling toward the demarc from the customer equipment is carried on the M lead, and signaling from the demarc to the customer equipment is over the E lead.



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Figure 4-3. A 4-Wire with E&M Interface

## **CHAPTER 4 REVIEW QUESTIONS**

1. Sketch a 2-wire interface.
2. Sketch a 4-wire interface with Type I E&M signaling. Show all signaling and transmission directions.



## CHAPTER 5 TRANSMISSION TESTING

### INTRODUCTION

This chapter contains a discussion of the methodology used in testing circuits.

Many manufacturers build test equipment that is suitable for conducting the tests described in this chapter. The Bell System Technical Reference #41009 details the required physical and electrical characteristics of test equipment.

In this chapter, testing (with the exception of balance testing) is described with reference to the Hewlett-Packard Model 3551A Transmission Test Set. Balance testing is described with reference to the Wiltron Model 9041 Transmission Level and Return Loss Measuring Set.

Testing methodology is presented in terms of demarc-to-demarc tests.

Tests are presented in the sequence in which they would be performed in actual practice. The following tests are discussed in detail:

- Actual Measured Loss
- Slope
- Idle Channel Noise
- C-notched Noise
- Balance

### ACTUAL MEASURED LOSS

The purpose of this test is to obtain the Actual Measured Loss (AML) of the circuit being tested. The test described here is for the far-to-near direction; testing in the near-to-far direction is similar.

**TEST CONFIGURATION:** The facility to be tested and the configuration used to describe testing is shown in Figure 5-1. The facility shown and the TLPs given are arbitrary and do not necessarily represent a "real-world" situation. Actual situations will vary in detail; however, the basic testing methodology presented here has general application.

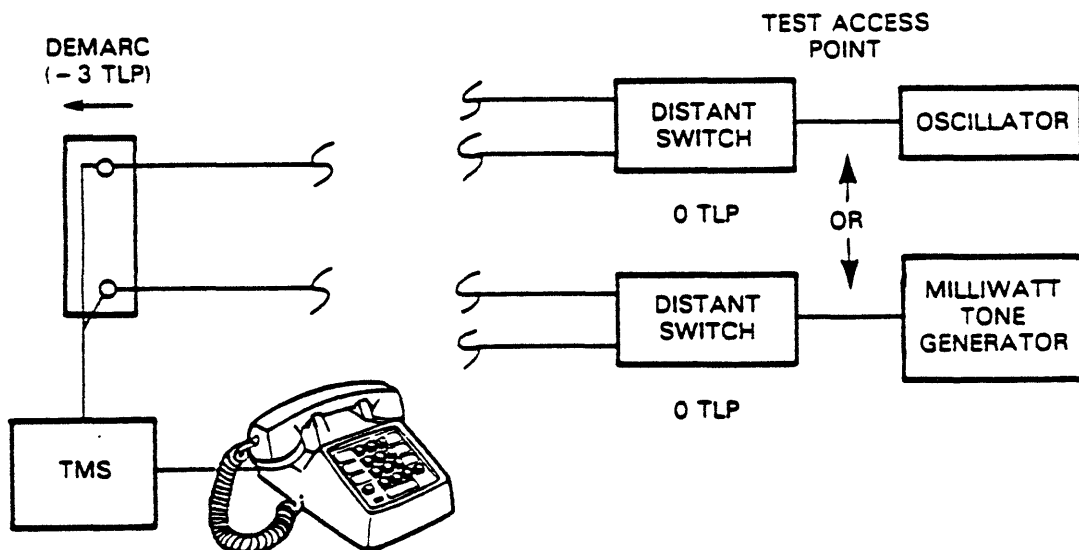


Figure 5-1. Test Configuration

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## Part II Transmission

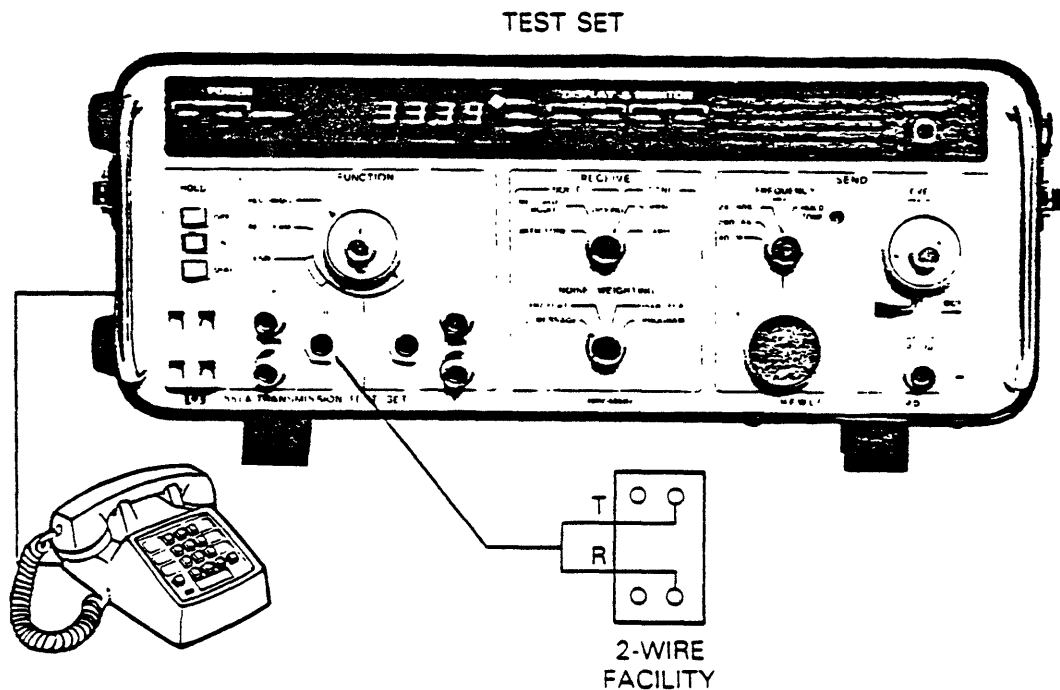
Coordination with the distant end will be required in order to conduct the test. As described here, the test requires a technician located at the distant test access point. As indicated in Figure 5-1, an alternative setup using a 1004 Hz tone generator (milliwatt) that can be directly dialed may be used. A telephone number that connects to either a test access point or a milliwatt tone must be obtained.

**TEST PROCEDURES:** Testing may begin when connection with the distant end has been established.

The test equipment is set up as shown in Figure 5-2. Although the equipment referenced here is the HP 3551A, tests using different equipment are procedurally similar.

(It is critical that the selected impedance be the same as the terminating impedance, otherwise, all test measurements will be in error.)

Ask the tester at the distant end to apply 1004 Hz tone to the test access jack at 0 dBm. Record the level shown on the Test Set as the AML for the far-to-near direction.



### SWITCH

POWER  
 DISPLAY & MONITOR  
 HOLD  
 FUNCTION  
 IMP (Impedance):  
 RECEIVE  
 NOISE WEIGHTING  
 SEND - FREQUENCY Range Hz  
 SEND - LEVEL Range dBm.

### SETTING

On (AC) or BATTERY  
 RECEIVE - LEVEL  
 OFF or HOLD as required  
 REC TERM  
 600 or 900 as appropriate  
 TONE - NORMAL  
 C-MESSAGE  
 200-6K  
 0 dBm

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Figure 5-2. Initial Test Equipment Setup



**CALCULATING THE RESULTS:** For the circuit shown in Figure 5-1, the Expected Measured Loss (EML) is 3.0 dB because the transmit point is the 0 TLP and the receive point is the -3 TLP. Suppose that the reading on the Test Set is -3.2 dB, indicating an AML of +3.2 dB. Calculate the results as follows:

$$\begin{aligned} \text{EML} &= -3.0 \text{ dBm} \\ \text{AML} &= -(-3.2) \text{ dBm} \\ \text{AML Deviation} &= \frac{\quad}{\quad} +0.2 \text{ dB (more loss)} \end{aligned}$$

## SLOPE

The purpose of the slope test is to measure the attenuation distortion of the circuit.

**TEST PROCEDURES:** You may proceed directly to the slope test from the AML test. The test setup is the same as in Figure 5-2, except where changes in switch settings are called for.

Ask the tester at the distant end to send 1004 Hz at -13 dBm0. (In this case, -13dBm0 is -13dBm, since the test access point is the 0 TLP.)

The receive level should be 13 dB,  $\pm 0.5$  dB lower than the AML test. (In this case, the reading should fall between 15.7 dB and 16.7 dB.) A reading outside of these limits indicates that the circuit has gain non-linearities.

Record the reading as the 1004 Hz reference level.

Ask the tester to change the frequency from 1004 Hz to 404 Hz. Change the DISPLAY & MONITOR switch from RECEIVE - LEVEL to RECEIVE - FREQ and check the frequency. It should be approximately 404 Hz.

Change the DISPLAY & MONITOR switch back to RECEIVE - LEVEL and record the reading as the 404 Hz level.

Ask the tester to change the frequency from 404 Hz to 2804 Hz. Change the DISPLAY & MONITOR switch to RECEIVE - FREQ and check the frequency. It should be approximately 2804 Hz.

Change the DISPLAY & MONITOR switch back to RECEIVE - LEVEL and record the reading as the 2804 Hz level.

**CALCULATING THE RESULTS:** Suppose that the following readings were recorded for the test:

$$\begin{aligned} 1004 \text{ Hz Reference Level} &= -16.1 \text{ dBm} \\ 404 \text{ Hz Level} &= -16.0 \text{ dBm} \\ 2804 \text{ Hz Level} &= -17.0 \text{ dBm} \end{aligned}$$

Calculate the results as follows:

404 Hz Slope:

$$\begin{aligned} 1004 \text{ Hz Reference Level:} & -16.1 \text{ dBm} \\ 404 \text{ Hz Level:} & -16.0 \text{ dBm} \\ 404 \text{ Hz Slope} &= \frac{\quad}{\quad} -0.1 \text{ dB} \end{aligned}$$

2804 Hz Slope:

$$\begin{aligned} 1004 \text{ Hz Reference Level:} & -16.1 \text{ dBm} \\ 2804 \text{ Hz Level:} & -(-17.0) \text{ dBm} \\ 2804 \text{ Hz Slope} &= \frac{\quad}{\quad} +0.9 \text{ dB} \end{aligned}$$

A minus (-) sign in the result indicates less loss while a plus (+) sign indicates more loss.

## Part II Transmission

The slope of the circuit is the deviation from the 1004 Hz reference level

The acceptable limits for the slope are to be obtained from the circuit design specifications.

### IDLE CHANNEL NOISE

The purpose of this test is to measure the noise on the circuit in the absence of any signal; this noise is also called background noise.

You may proceed directly to this test from the slope test.

The test setup is the same as in Figure 5-2, except where changes in switch settings are called for

Check with the tester at the distant end to assure that no tone is being transmitted and that the circuit is properly terminated.

Be sure that the NOISE WEIGHTING switch is set to C-MESSAGE.

Change the RECEIVE mode switch from TONE - NORMAL to NOISE - MESSAGE CIRCUIT

The display will read in dBm units. Record the reading as idle channel noise, referenced to the 0 TLP (dBmC0).

The required limits for idle channel noise will be given in the circuit design specifications. In general, idle channel noise of less than 31 dBmC0 is acceptable for any circuit.

### C-NOTCHED NOISE

The purpose of this test is to obtain a measure of the ratio between the signal power and noise power present on a circuit.

**TEST PROCEDURES:** You may proceed directly to this test from the idle channel noise test. The test setup is the same as in Figure 5-2, except where changes in switch settings are called for.

Ask the tester at the distant end to send 1004 Hz at -13 dBm0 (Since the tone is being transmitted from the 0 TLP, the transmitted power is -13 dBm.)

Be sure that the RECEIVE mode switch is set to NOISE - MESSAGE CIRCUIT and the NOISE WEIGHTING switch is set to C-MESSAGE. Record the reading as signal plus noise. There are three components to the power represented by this reading: test tone power, noise power generated as a consequence of the test tone, and background noise.

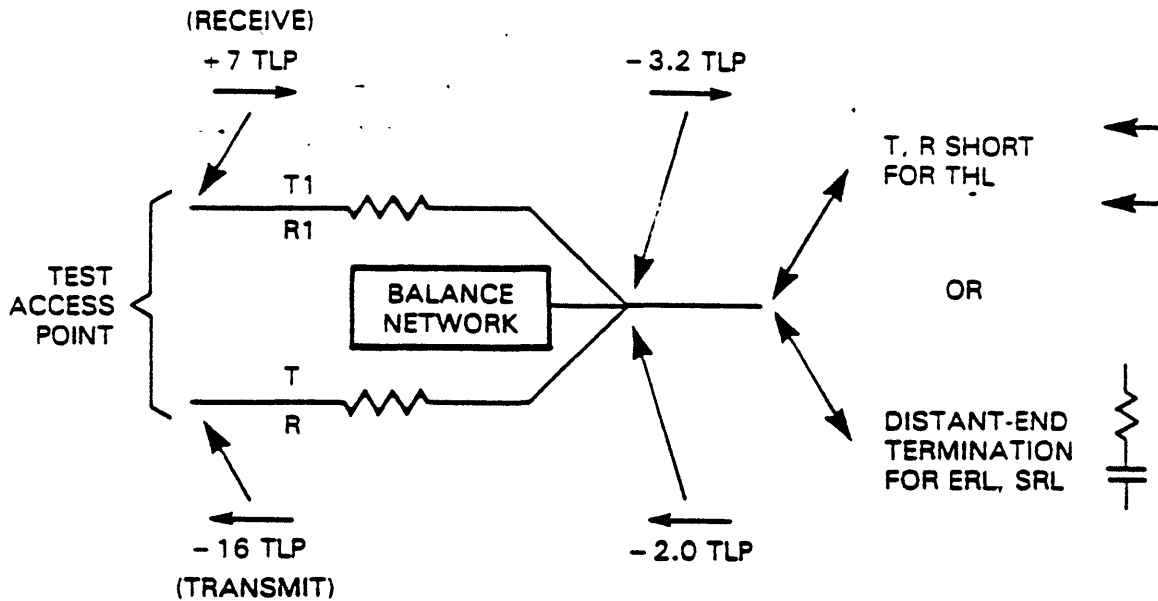
Change the RECEIVE mode switch from NOISE - MESSAGE CIRCUIT to NOISE - WITH TONE. This will cause the 1004 Hz test tone to be removed by a sharp notched filter. Record the reading as C-notched noise.

**CALCULATING THE RESULTS:** Suppose that the signal-plus noise measurement was 74 dBmC and the C-notched noise measurement was 50 dBmC. Calculate the signal-to-noise ratio as follows.

$$\begin{array}{rcl} \text{Signal plus noise} & = & 74 \text{ dBmC} \\ \text{C-notched noise} & = & \underline{-50 \text{ dBmC}} \\ \text{Signal-to-noise ratio} & = & 24 \text{ dBmC} \end{array}$$

### BALANCE

Figure 5-3 shows a typical hybrid that will be used to describe balance testing procedures. Although the test equipment shown is the Wiltron Model 9041 Transmission Level and Return Loss Measuring Set, tests using different equipment are similar.



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Figure 5-3. 4-Wire Termination Set

**CALCULATING THE THL:** Prior to conducting the balance tests, the expected value of the THL should be calculated. This is done by summing the gains and losses in the transmit and receive legs. One method of doing this is to use the TLP values for the circuit:

$$\begin{aligned} \text{Receive Path Loss} &= +7 - (-3.2) = 10.2 \\ \text{Transmit Path Loss} &= -2 - (-16.0) = 14.0 \\ \text{Expected THL} &= 10.2 + 14.0 = 24.2 \end{aligned}$$

The measured THL should be approximately this value. If it is not, there may be a problem with the hybrid or the test setup.

**MEASURING THE THL:** The equipment setup and initial switch settings for balance measurements are shown in Figure 5-4.

Note that there is a short between the 2-wire tip and ring leads.

The THL measurement is made by adjusting the ADD dB switches until black scale (bottom scale) of the meter reads 0. The total of all switch settings is the THL.

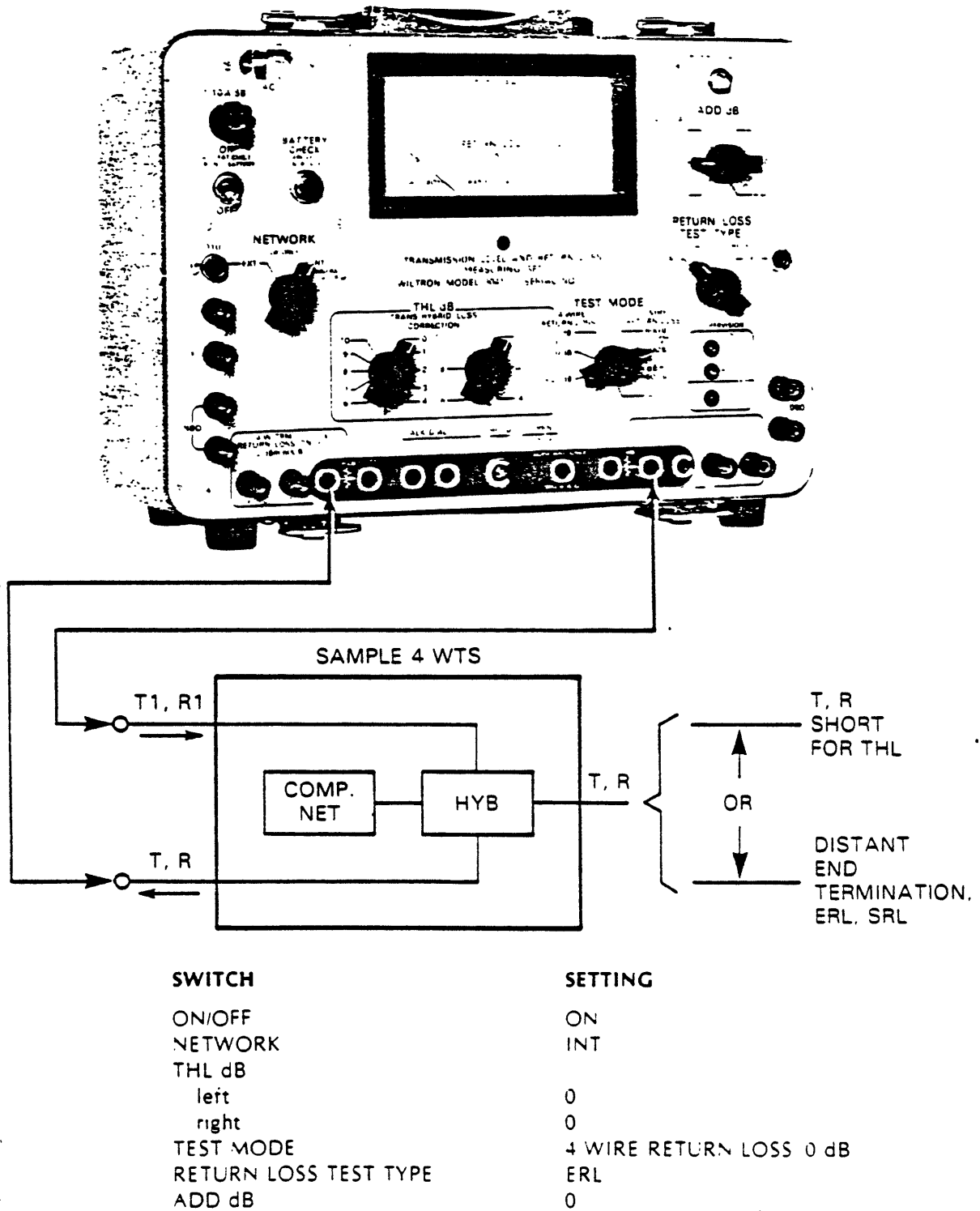
There are three switches that may be adjusted to zero the meter:

SWITCH	RANGE	INCREMENT
TEST MODE	0 - 20 dB	10 dB
THL dB (left)	0 - 10 dB	1 dB
THL dB (right)	0 - .8 dB	.2 dB

The most efficient way to zero the meter is to start with the switch that has the largest increment (TEST MODE) and proceed to the switch with the smallest increment (THL dB, right).

**Part II  
Transmission**

TEST SET (ILLUSTRATIVE ONLY)



IL 9201

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Figure S-4. Balance Test Setup

The previously calculated THL for the sample hybrid (24.2 dB) gives an indication of the start point for measuring the THL. Proceed as follows:

1. Set the TEST MODE switch to 20 dB. In this case, the meter now reads 3.2 dB
2. Set the THL dB (left) switch to 3 dB. The meter now reads .2 dB.
3. Set the THL dB (right) switch to .2 dB. The meter will now read 0 dB.

**CALCULATING THE RESULTS OF THE THL TEST:** The THL is equal to the sum of the switch settings, as follows:

SWITCH	VALUE
TEST MODE	20 dB
THL dB (left)	3 dB
THL dB (right)	<u>.2 dB</u>
THL	23.2 dB

The measured THL is close to the calculated THL, indicating that the test setup is correct and that the termination set is functioning as expected.

**MEASURING THE ERL:** The ERL test is similar to the THL test, except that the 2-wire short is removed and the 2-wire path is terminated in the proper impedance.

When the 2-wire path is terminated, the return loss will increase because most of the power that is sent into the receive leg is now absorbed by the 2-wire line facility and its terminating impedance.

To measure the ERL, proceed as follows:

1. Leave all switch settings as they were at the end of the THL test.
2. Adjust the ADD dB switch until an on-scale reading is obtained on the meter. For the sample termination set, the meter reads 3.5 dB with an ADD dB setting of 40 dB.

**CALCULATING THE RESULTS OF THE ERL TEST:** The ERL is calculated by adding the ADD dB switch setting to the meter reading, as follows:

ADD dB	40 dB
Meter reading	<u>3.5 dB</u>
ERL	43.5 dB

**MEASURING THE SRL:** Two measurements are taken in the SRL test: SRL LO and SRL HI.

SRL LO measures the return loss for noise that represents the low end of the voice frequency band. This noise has its power concentrated between 200 and 500 Hz.

SRL HI measures the return loss for noise that represents the high end of the voice frequency band, from approximately 2200 and 3400 Hz.

To measure the SRL, proceed as follows:

1. Leave all switch settings as they were at the end of the ERL test.
2. To measure SRL LO, change the RETURN LOSS TEST TYPE switch to SRL LO. Adjust the ADD dB switch until an on-scale reading is obtained on the meter. For the sample termination set, the meter reads 5.2 dB with the ADD dB switch set at 40 dB.
3. To measure the SRL HI, change the RETURN LOSS TEST TYPE switch to SRL HI. For the sample termination set, the meter reads 3.0 dB with the ADD dB switch set at 40 dB.

## Part II Transmission

**CALCULATING THE RESULTS OF THE SRL TEST:** Calculate the results of the SRL test as follows:

SRL LO:		
	ADD dB	40 dB
	Meter reading	<u>5.2 dB</u>
	SRL LO	45.2 dB

SRL HI:		
	ADD dB	40 dB
	Meter reading	<u>3.0 dB</u>
	SRL HI	43.0 dB

The SRL reading to record is the one that gives the lowest return loss, in this case SRL HI. This indicates the frequency band that has the poorest return loss.

**EVALUATING BALANCE TEST RESULTS:** The values of ERL and SRL represent a circuit's "safety margin" against echo and singing. The higher the values, the less likely the circuit is to suffer echo impairment.

In general, longer transmission pathways require higher values for ERL and SRL. This is reflected in various Bell System Documents:

TYPE OF CIRCUIT	ERL	SRL
4-Wire Tie Trunk	27 dB	20 dB
2-Wire Short Haul Tie Trunk	18 dB	10 dB

These objectives are given here *for illustrative purposes only*. Actual objectives and limits must be obtained from circuit design objectives.

**ADJUSTING BALANCE:** Echo is the result of impedance mismatch between the compromise network in the hybrid and the 2-wire circuit.

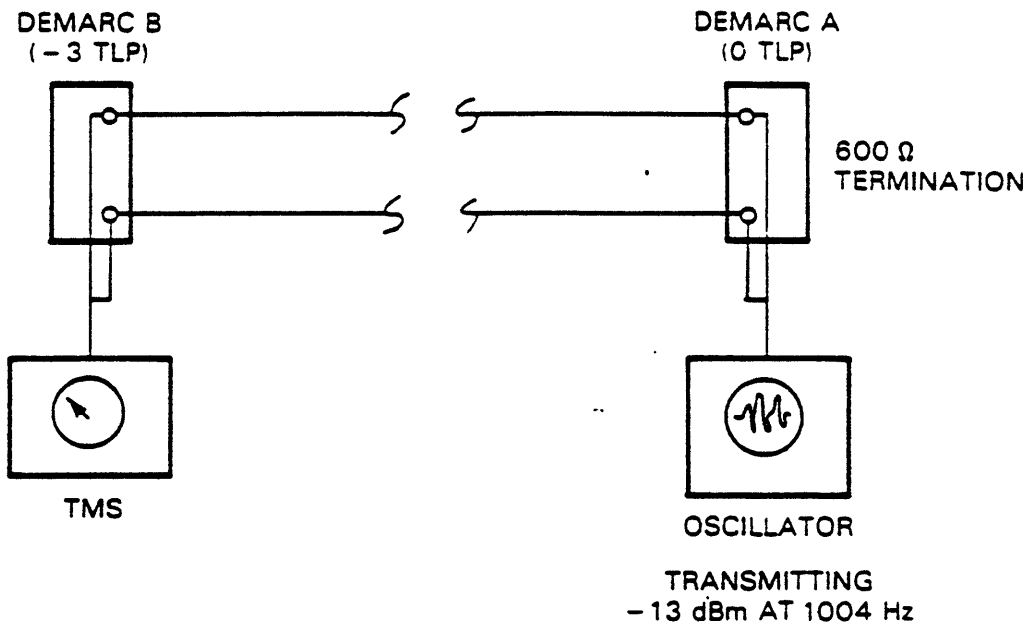
The impedance in the compromise network is usually made up of resistive and capacitive elements. Most hybrids contain internal compromise networks that may be adjusted by adding or removing *Network Build Out Capacitors (NBOC)*. Depending upon the construction of the hybrid, the adjustments are made by changing a screw-pot setting, changing switch settings or changing hard-wired straps.

There are cases where the impedance mismatch is too great to be compensated for by the internal NBOCs. These cases require the substitution of external compromise networks.

One type of external network is called a *Precision Balance Network (PBN)*; PBNs are commercially available.

In some cases, external networks may be engineered, constructed and installed in-house. Generally, these balance networks will employ both resistive and capacitive elements in order to "build out" the impedance.

## CHAPTER 5 REVIEW QUESTIONS

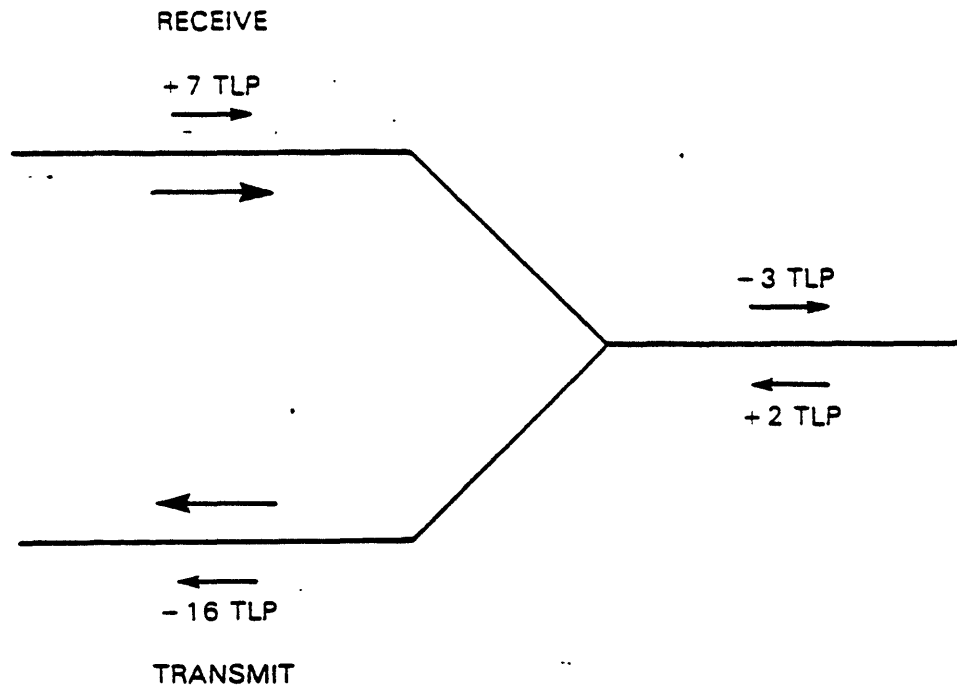


Answer the following questions with reference to the above figure.

1. What is the EML between A and B?
2. What is the expected receive level at B?
3. If a reading of -17.2 dBm is taken at B, what is the AML deviation?
4. The following measurements were taken. What is the slope?

404 Hz Reference Level = -17.0 dBm  
 1004 Hz Reference Level = -17.2 dBm  
 2804 Hz Reference Level = -18.0 dBm

**Part II  
Transmission**



Answer the following questions with reference to the above figure.

5. Calculate the THL that you would expect to see.
6. The following readings were recorded with the THL compensated for in the test equipment and the circuit properly terminated. What is the ERL?

ADD dB	30 dB
Meter reading	7.5 dB

7. The following readings were taken. What would you report as the SRL?

SRL LO:

ADD dB	= 30 dB
Meter reading	= 6 dB

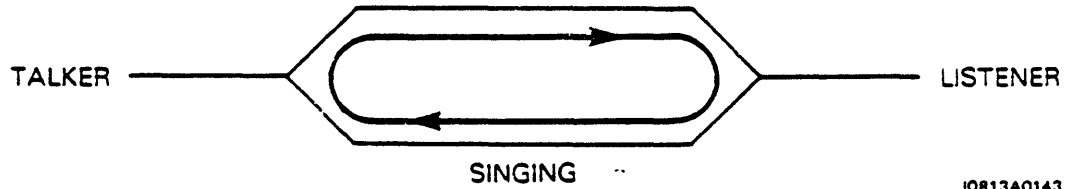
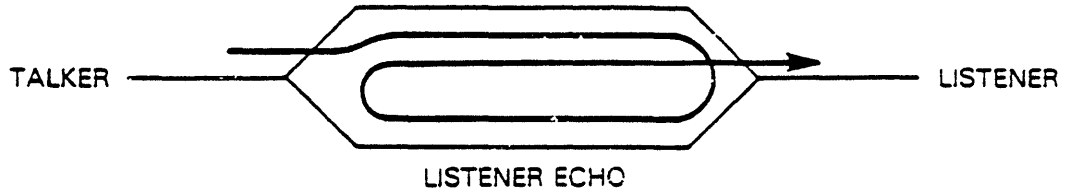
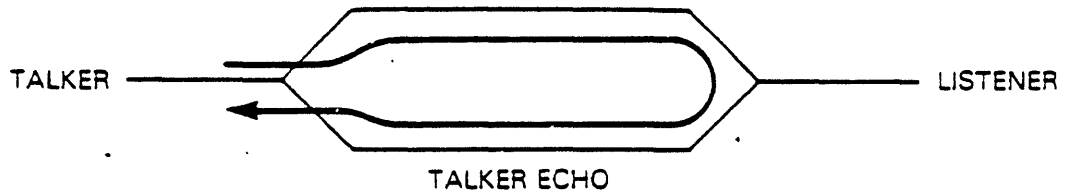
SRL HI:

ADD dB	= 30 dB
Meter reading	= 9 dB



**Part II  
Transmission**

6.



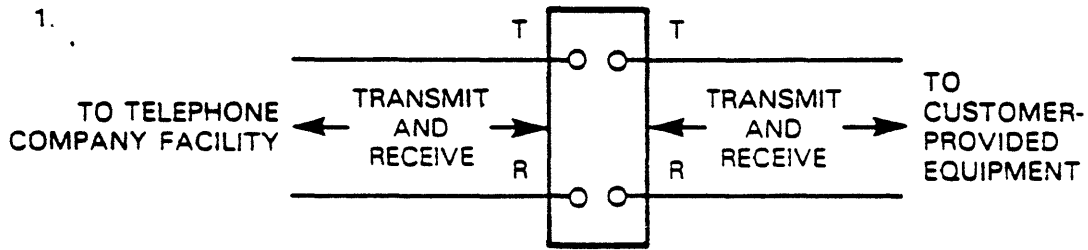
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7. It is a measure of the losses inherent in the 4-wire transmit and receive legs of the hybrid.

8. By matching the impedance of the 4 WTS Balance Network to the impedance of the 2-wire side.

**CHAPTER 4**

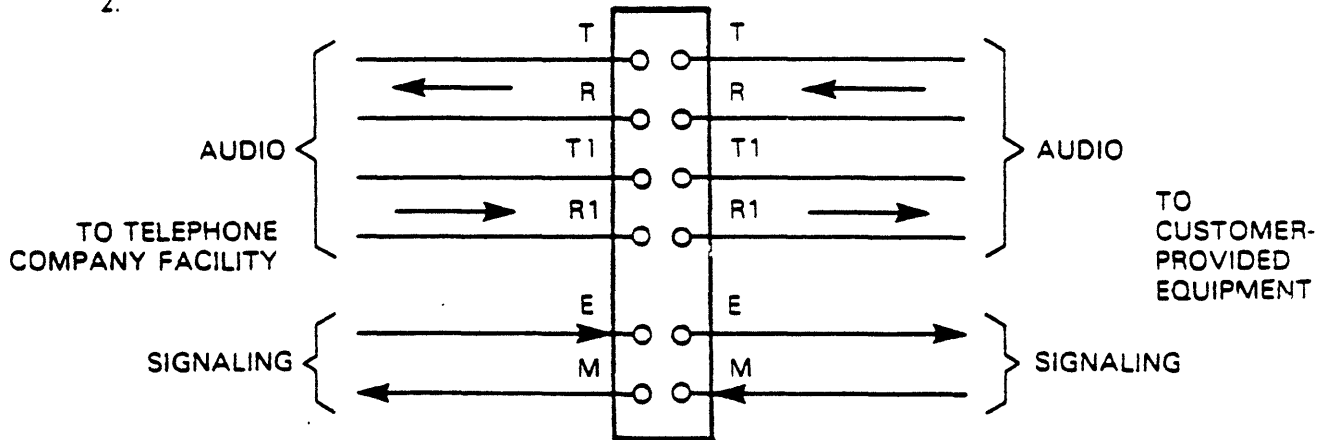
1.



2-Wire Interface

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2.



4-Wire with E&M Interface

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## APPENDIX A ANSWERS TO REVIEW QUESTIONS

### CHAPTER 1

1. Power and frequency.
2. Simple harmonic, sine curve.
3. By the addition of simple harmonic waves.
4. 0 to 6000 Hz.
5. 0 to 2000 Hz.
6. The higher frequencies.
7. About 300 or 3000 Hz.

### CHAPTER 2

1. The power that is present in telephone circuits is small.
2. The dB is a convenient mathematical expression of power ratios.
3. No. For wide ranges of power, the loss in dB will remain constant.
4. A loss of one-half of the power.
5. Ten times the power.
6.  $\text{dBm} = 10 \log\left(\frac{\text{Power, measured in mW}}{1 \text{ mW}}\right)$
7. 7 dBm
8. -13 dBm
9.  $\text{dBm} - \text{Reference} = \text{dBm}$   
 $58 - 90 = -32 \text{ dBm}$
10. 42 dBmC
11. 0 dBm is defined as 1 mW of power impressed upon a  $600 \Omega$  impedance at a frequency of 1004 Hz.

### CHAPTER 3

1. AML Deviation = -0.5 dB (less loss).
2. -0.5 dB at 404 Hz, to +0.2 dB at 2804 Hz.
3. Equalizers.
4. By "masking" the desired signal or through distracting effects, such as with crosstalk.
5. S/N = 22 dB.



**CHAPTER 5**

1.  $-3$  dBm
2.  $-16$  dB
3.  $+1.2$  dB
4. The slope is  $+0.8$  dB at 2804 Hz,  $-0.2$  at 404 Hz.
5. Expected THL = 24 dB
6. ERL = 37.5 dB
7. SRL = 36 dB, the smaller of the two values.

## INDEX

Absolute Power	P.11/2-1	dBm	P.11/2-4
Active Condition (off-hook)	P.1/1-9	dBm	P.11/2-7, T.2-2
Actual Measured Loss	P.11/3-2, 5-1, F.5-1 and 5-2	dBmC	P.11/2-7
Address Signaling, outpulsing methods used in		dBmCO	P.11/2-8
Senderized operation	P.1/1-14	DC Signaling Systems, types of	
Non-senderized operation	P.1/1-14	Loop Signaling	P.1/2-4
Address Signaling, types of		E&M Signaling	P.1/2-4
Dial Pulsing	P.1/1-9, F.1-9	Duplex Signaling	P.1/2-4
Dual Tone Multifrequency (DTMF)	P.1/1-10, T.1-1, 2-5	Decibel	P.11/2-3, T.2-1
Multifrequency Pulsing	P.1/1-11, T.1-2, 2-5	Delay Dialing	P.1/5-1, F.5-1a and 5-1b
Common Channel Interoffice	P.1/1-12, F.1-10	Delay Distortion	P.11/3-4
Alerting Signals, types of		Demarcation Point	P.1/2-5, F.2-4
Ringing	P.1/1-13	Dial Pulsing	P.1/1-9, F.1-9
Flashing	P.1/1-13	Dial Tone	P.1/1-13
Re-ringing	P.1/1-13	DID Trunks	P.1/1-8
Re-call	P.1/1-13	Direct Inward Dialing (DID)	
Receiver off-hook	P.1/1-13	Overview of	P.1/6-1, F.6-1
Alternate Route Choice	P.1/1-15	Detailed operation of	P.1/6-2, F.6-2 thru 6-4
Amplitude	P.11/1-1	Overall operation of (with schematics)	P.1/App.C
Analog Carrier Transmission Facilities	P.1/2-3	Distortion, Signal	
Analog Facility Terminal (AFT)	P.1/2-9, 3-7, 4-8	Attenuation	P.11/3-3
Answers to Review Questions	P.1/App.D; P.11, App.A	Frequency Shift	P.11/3-4
Answer Supervision	P.1/1-16	Delay	P.11/3-4
Attenuation-Distortion	P.11/3-4	Dual Tone Multifrequency (DTMF)	P.1/1-10, T.1-1, 2-5
Audible Ring	P.1/1-13	Duplex Signaling	P.1/2-4
Balance, adjustment of	P.11/5-8	E&M Signaling	P.1/2-4, 2-6, T.2-2, F.2-6 thru' 2-9
Balance Testing	P.11/3-10, 5-8	Type I	P.1/2-8, F.2-7a
Balance Theory	P.11/3-9, F.3-6	Type II	P.1/2-8, F.2-7b
Call Clean-up	P.1/1-16	E&M Signaling Interface	P.1/2-6, F.2-7a and 2-7b, T.2-2, P.11/4-1, F.4-2, 4-2, F.4-3
Call Origination	P.1/1-14	Echo	P.11/3-7
Call Routing	P.1/1-14	Echo, types of	
Change of State, end to end	P.1/2-9, T.2-3, F.2-8	Talker Echo	P.11/3-8, F.3-5
Circuit Operation	P.1/Ch.2	Listener Echo	P.11/3-8, F.3-5
Circuits, types of		Singing	P.11/3-8, F.3-5
Line	P.1/1-1, 1-6	Echo Return Loss (ERL)	P.11/3-10
Trunk	P.1/1-1, 1-4, 1-6	End to End Change of State	P.1/2-9, T.2-3, F.2-8
Shared	P.1/1-4	ERL Test	P.11/5-7
C-Message Weighting	P.11/2-7, F.2-4	Estimated Measured Loss (EML)	P.11/3-2
C-Notched Noise Test	P.11/5-4	Flashing	P.1/1-13
Common Channel Interoffice Signaling (CCIS)	P.1/1-12, F.1-10	Four-Wire Metallic Transmission Facilities	P.1/2-1, F.2-3
Complex Sounds	P.11/1-3	Four-Wire Termination Set	P.1/2-6, F.2-5, P.11/3-7, F.3-4
Complex Waveforms	P.11/1-3, F.1-2, F.1-3	Four-Wire with E&M Interface	P.11/4-2, F.4-3
Computerized Branch Exchange (CBX)	P.1/1-6	Frequency Shift Distortion	P.11/3-4
Controlled Outpulsing	P.1/Ch.5	FX Trunks	P.1/1-8
Conventions and Symbols	P.1/App.A, App.B, App.C		
CO Trunks	P.1/1-8		
Crosstalk	P.11/3-7		

## INDEX (Continued)

Clare	P 1/2-10, 3-1, 5-5, F.5-3	Multifrequency Pulsing	P 1/1-11 T 1-2, 2-5
Ground Start Signaling		Multiplexing	P 1/2-3
Overview of	P 1/4-1, F 4-1, T 4-1	Network Build Out Capacitors	P 1/5-8
Detailed operation of	P 1/4-3, F 4-2 thru' 4-9	Networks, overview of	P 1/Ch 1
Overall operation of (with schematics)	P.1/App.B	Network, types of	
Ground Start to E&M Interface		Simple	P 1/1-1, F 1-1
Detailed operation of	P 1/4-8, F 4-10a thru' 4-12b	Six-point Mesh	P 1/1-2, F 1-2
Overall operation of (with schematics)	P.1/App.B	Star	P 1/1-3, F 1-3
Hearing, characteristics of	P.11/Ch.1	Three-Switch	P 1.1-4 F 1-6
Hertz	P.11/2-1	Two-Switch	P 1/1-4, F 1-5
Hybrid Circuit	P.1/2-6	Noise, measurement units of	
Idle Channel Noise	P.11/3-5, F.3-2	dBrnC	P 11/2-7 T 2-2
Idle Channel Noise Test	P 11/5-4	dBrnC	P 11/2-7
Idle Condition (on-hook)	P 1/1-9	dBrnC0	P 11/2-8
Impulse Noise	P 11/3-7	Noise, types of	
In-Band Tone Signaling System	P 1/2-5	Idle Channel Noise	P 11/3-5, F 3-2, 5-4
Information Signaling	P 1/1-12	Signal-Dependent Noise	P 11/3-6
Information Signals, types of		Impulse Noise	P 11/3-7
Dial Tone	P.1/1-12	Crosstalk	P 11/3-7
Audible Ring	P.1/1-12	Non-Senderized Operation	P 1/1-14
Line Busy	P 1/1-12	OPX Lines	P 1/1-9
Reorder	P 1/1-12	Out-of-Band Tone Signaling System	P 1/2-5
Recorded Announcement	P.1/1-12	Outpulsing, controlled methods of	P 1/Ch.5
Interface		Precision Balance Network	P 11/5-8
Demarcation Point	P.1/2-5, F.2-4	Private Branch Exchange (PBX)	P 1/1-1, 1-3, 1-7
Four-Wire Termination Set	P 1/2-6, F.2-5; P 11/3-7, F 3-4	Recorded Announcements	P 1/1-13
E&M Signaling Interface	P.1/2-6, F.2-7a and 2-7b, T.2-2; P.11/4-1, F 4-2, 4-2, F 4-3	Reorder	P 1/1-13
Line Busy	P 1/1-13	Relative Loss	P 11/2-2
Line Circuits	P 1/1-1, 1-6	Relative Power	P 11/2-1
Listener Echo	P 11/3-8, F 3-5	Review Question Answers	P 1, App D, P 11, App A
Loop Signaling	P 1/2-4	Senderized Operation	P 1/1-14
Loop Start Signaling		Shared Circuits	P 1/1-4
Overview of	P 1/3-1, F.3-1, T 4-1	Signal-Dependent Noise	P 11/3-6
Detailed operation of	P 1/3-3, F 3-2 thru' 3-7	Signal Distortion	
Overall operation of (with schematics)	P.1/App.A	Attenuation Distortion	P 11/3-3
Loop Start to E&M Interface		Frequency Shift Distortion	P 11/3-4
Detailed operation of	P.1/3-6, F 3-8a thru' 3-9b	Delay Distortion	P 11/3-4
Overall operation of (with schematics)	P 1/App.A	Signaling Formats	
Loss, transmission	P 11/3-1, F 3-1	Loop Start	P 1/2-10 Ch 3, App A
Measurement Units, in transmission	P 11/Ch.2	Ground Start	P 1/2-10, Ch 4, App B
Metallic Facility Terminal (MFT)	P 1/3-7, 4-8	Delay Dialing	P 1/2-13, F 2-9a, Ch 5
Metallic Transmission Facilities		Wink Start	P 1/2-14, F 2-9b, Ch 5
Two-Wire	P 1/2-1, F.2-2	Direct Inward Dialing	P 1/2-14, Ch.6, App C
Four-Wire	P 1/2-1, F 2-3	Signaling Systems, types of	
Milliwatt	P 11/2-1	DC	P 1/2-4
Modulation	P 1/2-3	Tone	P 1/2-4
		Signaling, types of	
		Supervisory	P 1/1-9
		Address	P 1/1-9
		Information	P 1/1-12
		Alerting	P 1.1-13
		Test	P 1/1-14
		Simple Harmonic Waveforms	P 11/1-3 F 1-2

## INDEX (Continued)

Simple Network	P.1/1-1, F.1-1	WATS Trunks	P.1/1-8, F.1-8
Simplexing	P.1/2-10	Tie Trunks	P.1/1-8
Singing	P.11/3-8, F.3-5	DID Trunks	P.1/1-8
Singing Return Loss (SRL)	P.11/3-10	OPX Lines	P.1/1-9
Single-Frequency (SF) Signaling	P.1/2-5	Two-Switch Network	P.1/1-4, F.1-6
Single-Frequency Return Loss	P.11/3-10	Two-Wire Interface	P.11/4-1, F.4-1
Six-Point Mesh Network	P.1/1-2, F.1-2	Two-Wire Metallic Transmission Facilities	P.1/2-1, F.2-2
Slope	P.11/5-3	Two-Wire with E&M Interface	P.11/4-1, F.4-2
Sound	P.11/1-1	Voice Frequency Band	P.11/1-6
Sounds complex	P.11/1-3, F.1-2	WATS Trunks	P.1/1-8, F.1-8
Sound, Waves	P.11/1-1, F.1-1	Watt	P.11/2-1
Speech	P.11/Ch.1	Waveforms	
Speech, characteristics of	P.11/Ch.1	Simple harmonic	P.11/1-3, F.1-2
SRL Test	P.11/5-7	Complex	P.11/1-3, F.1-2 and 1-3
Star Network	P.1/1-3, F.1-3	Wink Start	P.1/5-3, F.5-2a and 5-2b
Supervisory Signaling	P.1/1-9		
Supervisory States			
Idle	P.1/1-9		
Active	P.1/1-9		
Switchboard, manually operated	P.1/1-4, F.1-4		
Symbols and Conventions	P.1/App. A, App. B, App. C		
Talker Echo	P.11/3-8, F.3-5		
Telco Interfaces	P.11/Ch.4		
Telephone Call, typical phases of			
Call Origination	P.1/1-14		
Call Routing	P.1/1-14		
Answer Supervision	P.1/1-16		
Call Disconnect	P.1/1-16		
Call Clean-Up	P.1/1-16		
Test Signals	P.1/1-14		
THL Test	P.11/5-5		
Three-Switch Network	P.1/1-4, F.1-6		
Tie Trunks	P.1/1-8		
Tone Signaling Systems, types of			
In-Band	P.1/2-5		
Out-of-Band	P.1/2-5		
Trans-Hybrid Loss (THL)	P.11/3-10		
Transmission			
Loss of	P.11/3-1, F.31		
Measurement units of	P.11/Ch.2		
Transmission Facilities			
Metallic	P.1/2-1		
Analog Carrier	P.1/2-3		
Transmission Level Point (TLP)	P.11/2-4, F.2-2		
Performance Parameters	P.11/Ch.3		
Testing	P.11/Ch.5		
Units of Measurement	P.11/2-1		
Transmission Test Procedures			
Actual Measured Loss	P.11/5-1, F.5-1 and 5-2		
Slope	P.11/5-3		
Idle Channel Noise	P.11/5-4		
C-Notched Noise	P.11/5-4		
Balance	P.11/5-4		
Trunk Circuits	P.1/1-1, 1-4, 1-6		
Trunks			
CO Trunks	P.1/1-8		
FX Trunks	P.1/1-8		





**TRUNKING AND TRANSMISSION  
FINAL EXAM**



## FINAL EXAM

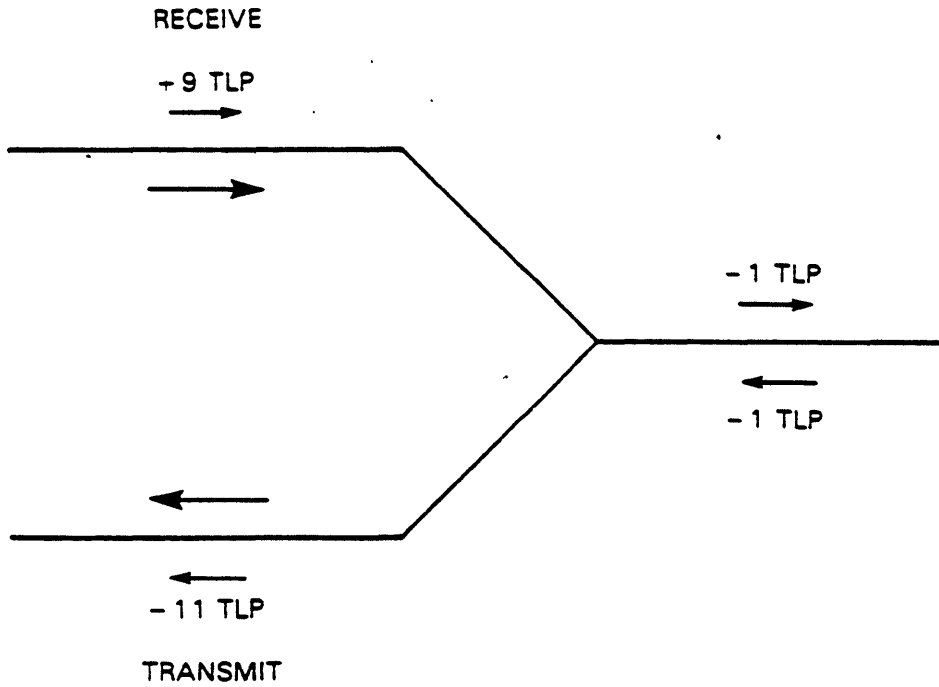
1. Can outgoing calls be made over DID trunks?
2. Calculate the THL that you would expect to see in Figure A?
3. What are the two supervisory conditions?
4. How does the PBX signal answer supervision to the CO on DID trunks?
5. Is the loss (or gain) in a circuit, when expressed in dB, dependent upon the absolute power carried by the circuit?
6. The following measurements were taken. What is S/N?

Noise with tone = 72 dBmC  
C-notched noise = 36 dBmC

7. Define glare.
8. Define percent break.
9. How does the CO respond to a distant-end disconnect on ground start trunks?
10. If a power measurement of +8 dBm is taken at the +3 TLP, what would the power measurement be at the 0 TLP?
11. If a reading of -21.9 dB is taken at B, what is the AML deviation? Refer to Figure B.
12. What is the function of 4-wire termination sets?
13. The following measurements were taken. What is the slope?

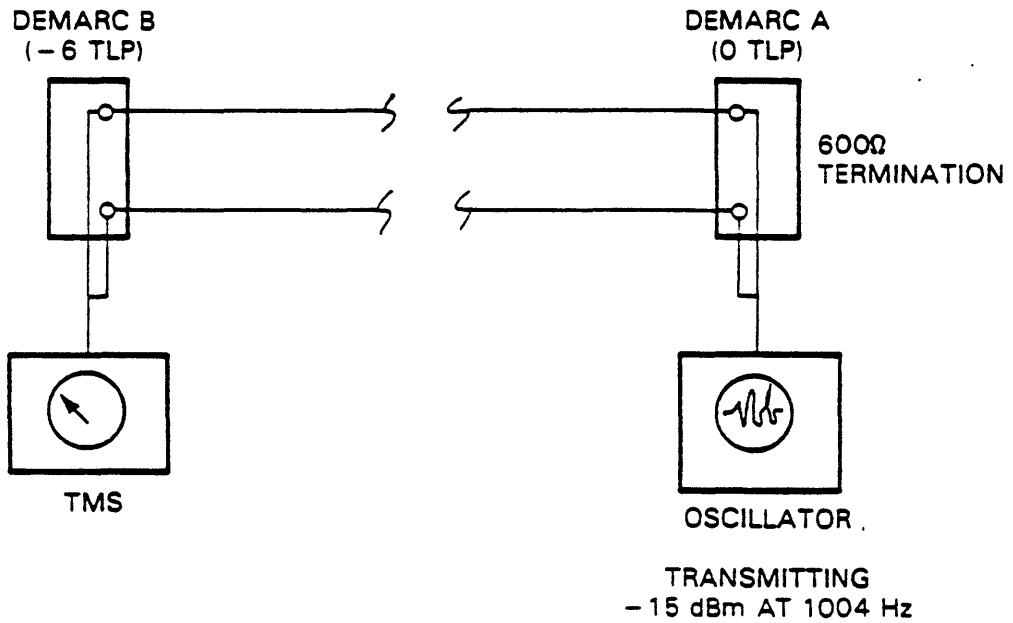
404 Hz Reference Level = -11.8 dBm  
1004 Hz Reference Level = -13.0 dBm  
2804 Hz Reference Level = -13.8 dBm

14. In delay dial operation, what is the start dialing signal?
15. What is the purpose of OPX lines?
16. Define dBm.
17. What is the purpose of CO trunks?
18. What is the difference between a line circuit and a trunk circuit?
19. What is the frequency range of the human voice?
20. What are two methods of controlled signaling?
21. In wink start, what is the start dialing signal?
22. What is the EML between A and B in Figure B?



10813A0145

Figure A



10813A0144

Figure B

23. The following measurements were taken. What is the AML deviation?

$$\begin{aligned} \text{EML} &= -3.2 \text{ dBm} \\ \text{AML} &= \underline{-5.6 \text{ dBm}} \\ \text{AML deviation} &= ? \end{aligned}$$

24. The following readings were taken. What would you report as the SRL?

SRL LO:

ADD dB	20 dB
Meter reading	2.4 dB

SRL HI:

ADD dB	10 dB
Meter reading	9.6 dB

25. Sketch a 4-wire interface with Type 1 E&M signaling. Show all signaling and transmission directions.

26. What are the electrical states of the tip and ring leads in the idle condition in loop start operation:

At the CO?  
At the PBX?

27. What are the phases of a telephone call?

28. Why must SRL be controlled?

29. Why is ground start signaling superior to loop start for senderized operation?

30. If the end-to-end noise objective for a circuit is 27 dB<sub>BrnC0</sub>, what would the maximum acceptable reading be at a +5 TLP?

31. Which method of controlled outpulsing provides better glare protection?

32. What are the electrical states of the tip and ring leads in the idle condition for ground start.

At the CO?  
At the PBX?

33. If a 0 dBm 1004 Hz test tone is applied at the 0 TLP, what would the power measurement be at the -5 TLP?

34. Why is ground start signaling superior to loop start with respect to CO origination?

35. What is the PBX start dialing signal toward the CO for DID trunks?

36. What is the transmission bandwidth of telephone circuits?

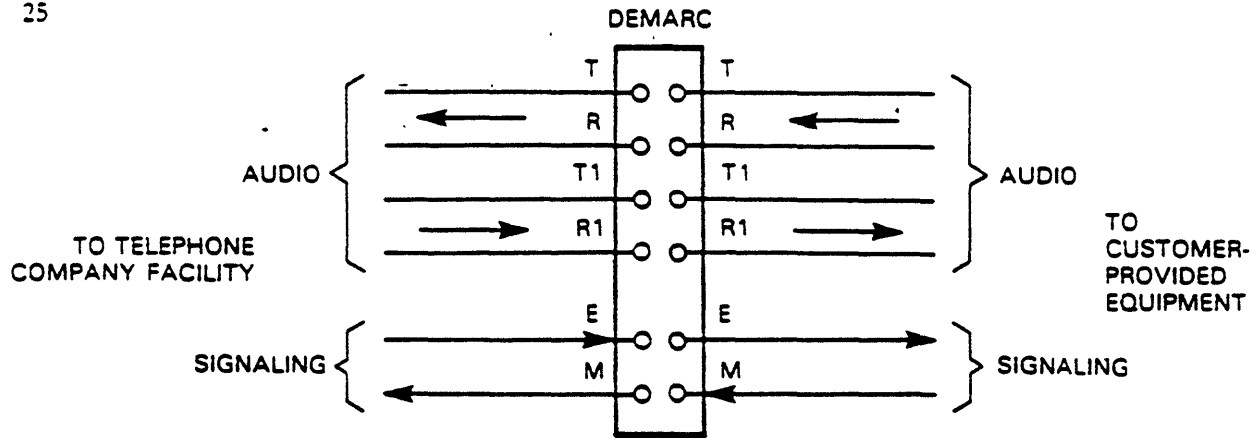
37. What type of trunk is used to connect PBXs?

38. What is the expected receive level at B in Figure B?

39. Give the transmission direction conventions for 4-wire circuits

40. What are the two outpulsing methods used in address signaling?

**ANSWER KEY  
TO  
TRUNKING AND TRANSMISSION  
FINAL EXAM**



IO813A0146

	<b>Tip</b>	<b>Ring</b>
26. CO	ground	battery
PBX	open	open

27. Origination, routing, answer supervision, disconnect and clean-up.

28. To avoid singing.

29. Tip ground from the CO provides the PBX with a positive start dialing signal, reduces glare and allows the CO to inform the PBX of a far end disconnect.

30. 22 dBrnC

31. Wink start.

	<b>Tip</b>	<b>Ring</b>
32. CO	open	battery
PBX	battery	open

33. -5 dBm

34. The PBX recognizes the tip ground from the CO as incoming seized and will not seize the circuit for an outgoing call. Glare is minimized.

35. A brief tip-ring reversal, or "wink."

36. About 300 to 3000 Hz.

37. Tie trunks.

38. -21 dBm

39. T and R toward facility; T1 and R1 from facility.

40. Senderized and non-senderized.

## ANSWER KEY TO FINAL EXAM

1. No. DID trunks are one-way incoming.
2. Expected THL = 20 dB.
3. On-hook and off-hook.
4. By reversing the state of the tip and ring leads. The tip is connected to battery and the ring is grounded.
5. No. For wide ranges of power, the loss in dB will remain constant.
6.  $S/N = 36$  dB.
7. Glare is the simultaneous seizure of both ends of a trunk.
8. Percent Break =  $100 \times \frac{\text{Break duration}}{\text{Pulsing period}}$
9. Removes the tip ground.
10. +5 dBm
11. 0.9 dB (The transmit tone is -15 dBm0.)
12. To convert from 2-wire to 4-wire transmission.
13. The slope is -1.2 dB at 404 Hz, +0.8 dB at 2804 Hz.
14. The transition from off-hook to on-hook at the far end switch.
15. To provide the convenience of extension dialing to remotely located telephones.
16.  $\text{dBm} = 10 \log \left( \frac{\text{Power, measured in mW}}{1 \text{ mW}} \right)$
17. To connect private networks to the public switched network.
18. Line circuits are dedicated to a single station set; trunk circuits connect switching systems and are shared among users.
19. 0 to 6000 Hz.
20. Wink start and delay dialing.
21. An on-hook to off-hook to on-hook "wink."
22. -6 dBm
23. 2.4 dB
24. SRL = 19.6, the smaller of the two values.