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MERLIN LEGEND™ COMMUNICATIONS SYSTEM APPLICATION NOTES

MERLIN LEGEND™ COMMUNICATIONS SYSTEM
Applications Note On Basic Trunking Concepts

Abstract

This Application Note describes the various types of trunks that link the MERLIN LEGEND Communications System with the telephone network. Operating characteristics of the various trunks are shown to be a function of both the trunk type and the capabilities of the LEGEND system hardware and software. The Note is designed to help Account Executives and System Consultants, so a basic approach to the subject is used. The concepts covered also apply to other customer premises switching equipment as well as the MERLIN LEGEND CS.

This Application Note is designed for use as a reference manual.
Refer to it each time you get involved with a MERLIN LEGEND system sale requiring a mixture of the various types of lines and trunks.

555-600-736

Jim Pastorius

Kevin Lyons

Writers/Editors

Contributors:

A. Cohen
D. Guerro
V. Illuzzi
R.G. Koppenheffer
D. Margolis
S.W. Osborne
H.T. Reeve
M. Stevenson
B. Tannu
C.A. White
J. Webb

MERLIN LEGEND TRUNKING CONCEPTS
APPLICATION NOTE INDEX

INTRODUCTION TO TRUNKING CONCEPTS

The way lines and trunks have been used to meet customers' needs over past decades is rapidly changing. T-1, ISDN technology, fiber optics, and other advances in telecommunications are forcing us to rethink what lines and trunks should be used for a PBX or Key system. It is critical, however. that everyone involved in giving customers the most advanced, yet economical. system have a basic understanding of the concepts which are the foundations for sending and receiving telephone voice and data messages.

The goal of this Application Note is to explain the various types of lines and trunks which may connect to the MERLIN LEGEND Communications System. The Note focuses on the concepts of each type of line or trunk. While it also covers major interactions between the LEGEND system and the various lines/trunks, it is not aimed at explaining everything needed to install, initialize. and maintain them on the system. This is fully covered in the LEGEND System Installation, Programming, and Maintenance Manual. Additional information on lines and trunks may be found in the LEGEND PBX and Key Systems Planning Guides, as well as the Systems Reference Manual.

TIP AND RING EXPLAINED

The terms "Tip" and "Ring" occur inevitably in any description of telephone lines and trunks.

These terms originated in the early days of telephony when telephone connections were made by an operator who inserted a plug into a jack. This plug was similar to the plug on a set of conventional stereo headphones in that it had three conductors. These conductors were the "Tip," corresponding to the tip of the plug. the "Sleeve," the longest Conductor at the base of the plug, and a ring of metal between the Tip and the Sleeve called "Ring."

The Sleeve was connected to a local electrical ground at the subscriber's premises and did not carry a signal to the telephone company Central Office (CO). The Tip and Ring conductors each connected to a wire that carried signals to the central office.

The terms became solidly embedded in telephone jargon and are still used in modem, electronic switching even though the actual "tips" and "rings" have long since disappeared. Today the terms Tip and Ring are used primarily as means for people in different places to identify precisely where each individual wire in each pair of wires (in a huge bundle of wires) needs to be connected. When a pair of wires is reserved for service as a trunk, one wire is designated Tip and the other wire is designated Ring.

During the process of ordering telephone service, a line assigner at the local operating company will reserve a pair of wires for each line or trunk ordered. Each individual wire in the bundle is identified by a unique color or combination of colors. Each trunk will have one color-coded wire assigned as the Tip lead and one color-coded wire assigned as the Ring lead of that particular trunk. For instance, a cable might contain one solid blue wire which would be reserved as the Ring lead on a given trunk, and one blue-and-white striped wire which would be reserved as the corresponding Tip lead for that same trunk. The line assigner will then furnish this information to installers at the customer's premises and technicians in the central office. In this way technicians can coordinate their installation work and make the proper connections between the customer's telephone equipment and the operating company's central office.

LINES AND TRUNKS

Telephone lines and telephone trunks are facilities that carry voice or data communications. They are similar in form and function, and the two terms are usually treated as if they are interchangeable. The fundamental difference between a line and a trunk is that a line connects a station instrument to a switching system, and a trunk connects one switching system to another switching system.

The connection between your home phone and the telephone company's central office is a line. The telephone facilities that serve key systems are also telephone lines, since a station in a mechanical key system accesses a telephone facility by pressing the specific button that corresponds to the exact facility desired. In a mechanical key system no switching takes place.

MERLIN LEGEND can be administered as a PBX, a telephone switching system that happens to be located on a customer's premises. A person using a LEGEND system telephone can access a wide variety of facilities, none of which need to be permanently connected to that specific voice terminal, by dialing a code (such as "9") and having the PBX select one facility from a group, or pool, of facilities. The voice connection from that station is then electronically switched to the selected facility.

Meanwhile, the facility that connects the voice terminal (station instrument) to the PBX is called a station line. In this example the voice terminal corresponds to your home phone, and the Legend system represents a scaled-down central office. Even though the station and the switching system are both in the same building, a facility that connects a station instrument with a switching system is a line.

Most of the facilities that connect the LEGEND system to the local central office are properly called trunks, but there are some misconceptions about this simple definition. Many PBXs, including MERLIN LEGEND, support Personal Lines. These facilities typically appear on a voice terminal button and pass transparently through the PBX, without being switched, to the central office. Selecting a personal line button on a voice terminal and lifting the handset brings dial tone directly from the central office.

Historically, most telephone operating companies automatically engineered a trunk to better standards than a line in a process called "conditioning." The central office equipment and the cable path used for trunks had a meet higher standards for transmission quality than equipment used for line. Today most operating companies are removing this provision from tariffs, and if a PBX requires conditioned trunks they will probably be available on an extra-cost basis.

LOOP-START TRUNKS: OPERATIONS

Loop-start facilities are the simplest and most common end-user facilities in the nation-wide telephone network. Loop-start facilities provide virtually no supervision between the central office and the customer premises equipment (CPE). For this reason, loop-start facilities are usually suited for use with telephone systems that provide human supervision. Loop-start facilities are primarily intended for single line sets, key systems with line status lights, or older PBXs. Loop-start facilities are generally not well suited for use with PBXs that provide mechanical or electronic supervision.

A loop-start line or trunk consists of two wires running from the Central Office to the Customer Premises Equipment (CPE). For historical reasons these wires are called Tip and Ring. (The term Ring, when applied to one of the leads in a line or trunk facility, does not refer to the ringing signal sent to announce an incoming call.) The CO applies battery voltage to the Ring lead and connects the Tip lead to ground. These wires meet at a switch in the CPE that is normally open, and are bridged by ringing detection equipment (such as the bell in a single line set) which provides a.high electrical resistance. [1]

Telephone equipment on a loop-start facility signals the central office that it needs the facility for an" outgoing call by closing this switch between the Tip and Ring leads. This may be as simple as lifting the handset from the switchhook if the equipment is a single line set. When this switch is closed the resistance between the 17p and Ring leads drops low enough to allow battery current to flow in a "loop" running from the CO out to the CPE, and back to the CO. When the CO detects current flowing through the loop it provides dial tone for an outgoing call.

The central office signals an incoming call on loop-start facilities by sending a ringing signal on the line. These are the only two signals that are uniformly used by loop-start facilities.

It is important to note that these signals do not give advance warning of a change in switchhook state. The only signal that the CPE sends to the central office that it is seizing a loop-start facility is the act of going off-hook on that line. The signal from the central office that alerts the CPE to an incoming call is the ringing signal received at the CPE at the same time *or slightly after* the call is connected to the facility.[2] Also, it should be noted, there is no defined method for a central office to terminate a call, so there is no reliable way to force a disconnect when only one of the two parties on a call hangs up.

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1. Actually, the proper term here is impedance, not resistance, but the effect is the same for all practical purposes, no current flows.

2. See *The Problem of Glare* in the next section.

LOOP-START TRUNKS: POTENTIAL-PROBLEMS

The most common application for loop-start facilities is single line residential service. Many of the disadvantages of a loop-start trunk in a PBX environment can be inferred from the operation of a loop-start residential telephone line, like the one you have at home.

The Problem of Glare

Perhaps, at least once in your life, you've picked up your home phone to make a call. only to hear a startled voice saying "hello?" instead of dial tone. This is called glare. Glare is the most obvious problem associated with loop-start facilities. This is due to the interaction of the extremely simple methods of signaling that a line is being seized.

When a call for a given loop-start facility comes in to a central office it is connected to the proper facility, and the signal from the central office ringing generators (90 volt, 20 cycle alternating current) is super imposed onto it. The central office ringing generators provide the ringing signal in a cycle of two-seconds-on (ringing), and four-seconds-off (silence). ringing signal to start at the reception of each individual call, so it is entirely possible for one call to be connected to a loop-start facility during the ringing period of the ringing cycle, and the next call to be connected during the silent period of the ringing cycle. An incoming call may be present on a loop-start facility for up to four seconds before ringing begins.

In a low traffic, single line, residential telephone situation this problem is uncommon, and sometimes it's even amusing. Even though the connection is made without warning in a residential situation there are only a few people that the incoming caller could be trying to reach. The problem, however, is more noticeable with higher traffic, multiline key systems. But since most key systems are relatively small and typically have close human supervision over line status and selection, these problems are usually identified and resolved quickly.

With a PBX, such as MERLIN LEGEND CS, the problem of glare becomes more serious and comples. Since ringing current is the only way that the central office can signal the PBX of an incoming call, and since the PBX typically asssumes that a facility is available until it receives the first cycle of ringing current, it's possible for the PBX to try to place an outgoing call on a loop-start trunk that has just been seized by the central office and is carrying an incoming call. The PBX thus connects, without waring, two parties who did not intend to reach each other.

A PBX trunk typically carries much higher levels of both incoming and outgoing traffic (more tails per hour) than a residential telephone line, so glare is statistically more likely to occur on a PBX trunk. Meanwhile, the ordinary station users on a PBX exercise little direct supervision over the individual trunks in the system.

On the simplest LEGEND system, one without Automatic Route Selection (ARS), a station user would typically lift the handset and hear dial tone from the PBX. The user would then dial an access code to a trunk or group of trunks, and wait for dial tone from the CO. It is at this point that the user might suddenly say "hello" to an unexpected incoming caller.

Features such as ARS magnify the problem. When the LEGEND system is administered for ARS the PBX is a "slenderized" system. It holds the digits that are dialed, selects the appropriate trunk, and then goes off-hook. LEGEND then waits two seconds on this trunk (under normal circumstances this is long enough to obtain dial tone from the CO), and then dials the telephone number. Notice that LEGEND doesn't actually recognize dial tone, and it cannot recognize glare, it just waits two seconds and dials.

The ARS feature also isolates the station user from the specific trunk, and even the group of trunks. that carries each outgoing call. Other features. such as Callback Queuing (LEGEND) or Busy-to-Idle (System 25) reminders, allow station users to seize trunks moments after a previous conversation concludes, and easily fast enough to beat a ringing signal on an incoming call.

The glare problem can also be compounded if the PBX customer has terminals or computers that place outgoing data calls without human supervision. If a terminal seizes a line for an outgoing call and encounters glare it will probably disconnect (hanging up on the incoming caller), record a line failure. and try again. The data user may become annoyed that a call attempt failed, but will never know about the rude reception that was just given to a potential customer.

Automated Attendant Ghost Calls and Loop-Start Facilities

Several customers with loop-start facilities and the Automated Attendant have reported problems with ghost calls. The human attendant who handles overflow calls from the Automated Attendant will notice that often, when the phone rings there will be no one there. or occasionally the attendant will pick up the ringing phone and hear dial tone.

The problem is caused by the fact that the Central Office does not send a reliable signal to force a disconnect at the telephone terminal when only one party on a loop-start line call hangs up. Until the second party hangs up, the loop-start line is considered in use by the CO, and can not be used to receive a new call. This happens sometimes with ground-start trunks as well, since there is usually 20-40 seconds for front end disconnect to pass through.

Consider the case of an automobile dealer that uses the Automated Attendant to direct calls to the new car sales, used car sales, parts, and service departments. During the course of the day, the alder will receive several calls from people who, for one reason or another, hang up while waiting for the Automated Attendant.

Since there is no positive disconnect on loop-start facilities, (the CO does not signal the set to hangup), the central office will continue to hold up the connection to the PBX. When this "connection" reaches the Automated Attendant, the usual list of dial-selectable destinations will be recited, but since there is no longer anyone on the call there will be no response. The Automated Attendant will time out and, in the absence of a positive response, it will assume that it is dealing with a caller who has a rotary dial phone. It will then forward the abandoned call to the human backup attendant. This person's phone will ring, but there won't be anyone there. The condition can also give the perception that the caller was cutoff.

LOOP-START TRUNKS:
WHEN TO USE LOOP-START TRUNKS

There are two times when you consider the use of loop-start facilities: when you *should* use loop-start facilities and when you *must* use loop-start facilities.

Proper Loop-Start Facility Applications

If loop-start facilities are prone to problems such as glare why are they still in use?

Most of the problems associated with loop-start facilities only come into play with automated PBXs. Loop-start facilities are perfectly acceptable for use with single-line telephones, key systems, and manual (cordboard or switchboard) attendant operated PBXs.[3] Some PBXs, especially those that cannot be administered by the customer, can operate properly with with loop-start facilities for one-way incoming and one-way outgoing trunks, that must be handled by an attendant.

Besides, the cost of converting all of the loop-start customer and CO equipment currently installed to ground-start operation would be an incremental expense.

Note: Loop-start trunks **are not recommended** for use with the MERLIN LEGEND configured for PBX mode operation. Ground-start trunks are *always* preferred.

Unavoidable Loop-Start Facility Applications

One situation that will require the use of loop-start facilities on the LEGEND system would be the connection of Centrex lines to the PBX. In some regions, Centrex is also called Essex or Centron lines. Most operating companies will provide Centrex service only on loop-start facilities. Some operating companies may consider converting the lines to ground-start facilities as an extra-cost special assembly, but this is at the operating company's option. Currently, the LEGEND system does not support the use of GS Centrex.

Consider the case of a state government. In the early seventies the state linked all departments, large and small, in one Centrex system in order to control inter-departmental message unit expenses. The individual departments terminate their Centrex lines in large, non-uniform key systems.

Now, departments like Motor Vehicles and the State Police, have grown to the size of small companies and occupy their own buildings. The Centrex is no longer capable of providing efficient internal communications for the large departments, and traffic demands from large departments are swamping service to smaller departments. The state would like to keep Centrex for convenient interdepartmental calling without message unit charges. The larger departments, however, need PBXs like the LEGEND system to take the load off the Centrex and for better internal communications. The local operating company will only provide Centrex service on loop-start facilities. In this case, some loop-start Centrex lines will terminate in the Legend system. Glare may occur on the Centrex lines. In general, **ground-start facilities are always preferred.**

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3. Some true manual PBXs are probably still present in the field. but you are not likely to encounter one during the installation of a modern PBX. On a true manual PBX all calls, both incoming and outgoing, must be handled by an attendant. there is no way to dial a code such as "9" for an outside line. h is this close supervision by the attendant that makes loop-start facilities acceptable for use with manual PBXs.

Cost

Unfortunately, the most common reason for using loop-start trunks is the least valid. This reason is cost.

If a customer currently has a key system, service at that location is undoubtedly supplied by loop-start lines. Converting the lines to ground-start trunks may involve a longer installation intend, and may incur additional charges from the operating company. As a result, many loop-start lines are simply re-used as loop-start trunks when the key system is replaced by a PBX.

However, if a MERLIN LEGEND CS is used, the customer has the flexibility of using either Loop Start or Ground Start lines/trunks. This advantage not only eliminates out-of-service conditions, as would be required if new replacement lines had to be installed, but can be a major cost-break by using the more economical Loop Start Modules. Also, the combined GS/LS board is a key advantage of the LEGEND system in facilitating the coordination with the telephone company in converting from LS to GS. It's not necessary to provide a loop-start module, and then have to replace it with a ground-start board should the lines be converted.

GROUND-START TRUNKS:
OPERATIONS

Ground-start facilities were specifically introduced to solve the problems that PBXs encounter on loop-start trunks. A ground-start facility provides an immediate signal when it is seized and it provides a positive signal when one party disconnects.

When a ground-start facility is idle the CO provides battery voltage on the Ring lead, but the CPE does not provide a ground on the Ring lead, and no current flows. Meanwhile, the CPE (LEGEND) monitors the Tip lead for ground at the CO, but the CO does not provide a ground on the Tip lead when the trunk is idle.

When the CO needs to seize a facility for an incoming call it selects an idle trunk, makes sure that the CPE has not applied ground to the Ring lead (no current is flowing), and the CO applies ground to the Tip lead. The ground connection on the Tip lead completes an electrical path, current begins to flow, and the CPE recognizes immediately that the facility has been seized for an incoming call. The CO also super imposes the ringing generators onto the facility, but ringing may not occur at once.

The significant operational difference here is that, unlike the signal from the ringing generators, the ground signal on the Tip lead is synchronized with the start of the incoming call. The CPE knows *immediately* that the facility is not available for outgoing traffic, even if it does not begin to ring for several seconds.

Ground-start facilities also provide a positive indication of a disconnection. When the distant party goes on-hook the CO removes battery from the Tip and Ring leads. Ground-start CPE (LEGEND) is designed to recognize this disconnect signal and remove its party from the facility. (This prevents the party from waiting for dial to to return, and prevent a toll restricted station from by-passing the restrictions.

Nearly all ground-start facilities are trunks, connecting PBXs with COs. There are only a few cases where round-start *lines* might be encountered. Some pay phones, especially older ones, use ground-start lines to obtain a positive disconnect signal and prevent a user from making multiple calls on a single coin. Also, some unattended data equipment uses ground-start lines so the modem can determine when to start dialing after going off-hook. Most modem equipment uses loop-start lines and alternate systems, such as a dial-tone detector in the modem, instead of ground-start lines.

GROUND-START TRUNKS:
POTENTIAL PROBLEMS

There are very few problems that can occur when properly functioning ground-start trunks serve a properly administered PBX, since modem PBXs and ground-start trunks were literally made for each other. This Application Note does not cover the kinds of problems that can occur when a trunk develops a problem and no longer functions properly. Suffice it to say that a malfunctioning ground-start trunk may behave just as poorly as a malfunctioning loop-start trunk, but a properly functioning ground-start trunk will always outperform a properly functioning loop-start trunk.

It's possible that when a large key system customer upgrades to a PBX, the conversion from loop-start lines to ground-start trunks may extend the installation interval. It is still always best to convert, and accept the one-time inconvenience, in order to enjoy a much better grade of service over the long haul.

The most common installation problems involved with ground-start trunks are reversed Tip and Ring connections and improper grounding of the PBX.

GROUND-START TRUNKS:
WHEN TO USE GROUND-START TRUNKS

Ground-start trunks should be used for a MERLIN LEGEND Communications System when configured as a PBX.

They offer the following major benefits:

1. The Tip ground seizure signal virtually eliminates the possibility that a two-way trunk can be seized from both ends simultaneously.

2. The "window of opportunity" for glare is reduced from four seconds to a small fraction of a second.

3. Ground-start operation also provides a positive indication of network disconnect when programmed at the CO for Calling Party Control, allowing a PBX to disconnect a station when the outside party on a call hangs up.

4. The possibility of a user circumventing toll restrictions is eliminated.

DIRECT INWARD DIALING (DID) TRUNKS: OPERATIONAL CHARACTERISTICS

Loop-start trunks and ground-start trunks are simple, two-way local facilities, i.e., they can handle incoming and outgoing calls. The operation of loop-start and ground-start trunks can be understood by analogy to the residential service in your your home.

Direct Inward Dialing (DID) trunks are different. **DID trunks are not two-way trunks and cannot be used for outgoing calls from the PBX, they are only used for incoming calls.** Also, an individual DID trunk is not associated on a one-to-one basis with a telephone number. Instead, a group of DID trunks is associated with a block of telephone numbers, and any DID trunk in the group can carry an incoming call directed to any DID number in the block.

DID trunks take advantage of the switching capabilities of both the PBX and the CO. The first step in an installation involving DID service is to order a block of DID telephone numbers from the operating company and the DID trunks that this group of numbers will serve. (**Note:** There will *always be* more DID numbers than DID trunks.) This block is a list of sequential numbers that will route incoming calls to the corresponding DID trunks. The minimum number of DID numbers that must be ordered varies from operating company to operating company. Some companies require a minimum of 20 DID numbers per block. Others may require as many as 100 numbers, and each DID number is billed to the customer.

When planning a DID setup, it should be kept in mind that generally each DID number corresponds to a particular telephone station number or hunt group. The number 555-8743, for example, may correspond to station 743, or to a Direct Call Group 555-7724 (Group code 724).

When the CO receives a call on a DID telephone number, it first identifies the block of telephone numbers that includes this DID telephone number. It then determines the group of DID trunks that this block of DID numbers has been administered to serve. The CO then selects one of these trunks and (in most cases) checks to make sure the trunk is operating properly. If the trunk is not operating properly, the CO selects the next available trunk in the same group and checks again. If no working trunks are available the CO routes the incoming call to a reorder signal (a fast busy signal).

When an idle trunk has been selected, the CO signals the PBX that it has an incoming DID call. Then the CO sends a series of pulses that identify the station that is intended to receive the call. (**Note:** In Controlled Address Signal Outpulsing DID systems, described below, the CO signals the PBX that it has an incoming DID call and waits for an acknowledgement from the PBX. When the PBX properly acknowledges the signal the CO sends the addressing information and the call proceeds.) The addressing information is taken from the network, and is based on the number that was actually dialed by the person who originated the call. This permits each user to have exclusive use of a telephone number without requiring a dedicated trunk.

Once a trunk has been seized -- the PBX has been alerted to the incoming DID call, and the addressing information has been sent -- there are three possible responses from the PBX.

a. The PBX will return an audible ringing signal if the desired voice terminal is available, if the call can be routed to a coverage position or attendant, or if the PBX can provide a recorded message (it will still ring first before providing the message). The calling party will not be charged unless the call is answered by a person or recorded message.

b. A busy signal is returned to the CO if the desired voice terminal is not available and the call cannot be routed to a coverage position or attendant, or if all DID trunks are in use.

c. A reorder signal is returned to the CO if the PBX cannot complete the call due to inadequate or faulty equipment, or if the PBX determines that the addressing information was incorrect or incomplete.

Calls to valid, but unassigned DID numbers, maybe routed to the PBX attendant.

DID Trunk Signaling Characteristics

The signaling associated with DID trunks is simplified a bit since DID trunks provide one-way incoming service only. The generic name for the type of signaling used by DID trunks is called Loop Reverse-Battery Signaling. The CO uses loop open (on-hook) and loop closed (off-hook) pulse signals to communicate with the PBX, and the PBX uses reversals of battery and ground, which reverses the flow of loop current on the Tip and Ring leads, to signal the CO.

The network controls whether current flows or doesn't flow, and the CPE detects the pulses of current.[4] The CPE controls the direction of the current flow, and the network detects the changes, or reversals, of current flow.

There are two different approaches to Loop Reverse-Battery Signaling: Immediate Address Signal Outpulsing and Controlled Address Signal Outpulsing. There are two varieties of Controlled Address Signal Outpulsing: Wink-start or Delay-dial.

Types of Loop Reverse-Battery Signaling		
Immediate Address Signal Outpulsing	Controlled Address Signal OutPulsing	
	Delay-dial	Wink-start

Immediate Address Signal Outpulsing is the simplest method of providing DID service. It is the only method used by the least sophisticated COs (DID-capable Step-by-Step Central Offices), and it may be used by any CO if the available equipment requires it. With Immediate Address Signal Outpulsing, when the CO must route an incoming DID call to the PBX it seizes a trunk by closing the loop and allowing current to flow. Then it immediately begins sending the pulses that carry the call addressing information without waiting for an acknowledgement signal from the PBX.

Controlled Address Signal OutPulsing provides better coordination between the CO and the PBX. When the CO must route an incoming DID call to the PBX using one of the Controlled Address Signal

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4. Notice that this is similar to the way that an analog voice terminal signals the PBX. When the voice terminal is on-hook (hung up) no current flows and the station line is idle. When the voice terminal goes off-hook the switchhook doses the loop formed by the Tip and Ring leads, current flows and the PBX prepares to accept a string of digits to dial for an external (or internal) call destination. In the case of DID, the network closes the loop and the PBX prepares to accept a suing of digits for an internal call destination.

Outpulsing methods, it seizes a trunk by closing the loop and allowing current to flow. When the PBX recognizes the trunk seizure from the CO it responds by reversing the battery and ground on the Tip and Ring leads. On delay-dial trunks this signal is called the delay-dial signal. On wink-start trunks this reversal, going from an on-hook condition to an off-hook condition and back again, is very brief (140 to 290 milliseconds) and it is called the wink signal. This handshaking is called an integrity check.

If the CO does not receive a proper response it will not route its incoming call over that particular DID trunk. In some wink-start systems the CO may be able to search for another available trunk and try to complete the call again (this is called retrial). In other wink-start systems, and in all delay-dial systems, retrial is not available. In these systems the incoming call will be routed to reorder tone at the CO and the failure will be recorded at the CO as a trouble. If enough integrity failures are recorded against a DID trunk it will be taken out of service by the operating company.

In general, the most efficient method for providing DID service is the wink-start type of Controlled Address Signal Outpulsing, but the particular variety of Loop Reverse-Battery Signaling that will be used for any given system is dictated by the capabilities of the CO, and will be decided by the operating company.

LEGEND OPERATION

Direct Inward Dialing calls ring on a System Access button (not "Line" buttons) on multibutton stations and are eligible for call coverage, forwarding, following, etc. DID calls can also ring directly into single-line stations, DGC groups, or QCC Consoles.

The system is capable of accepting 1 to 4 address digits over the DID trunks, for each call, as administered for each trunk group. The trunks are supported with the LEGEND system 800 DID module.

ADMINISTRATION

The use of DID trunks requires the following administration be completed:

- Send calls for unassigned DID numbers to a specified endpoint (default is station 10). This number can be a station extension, a DGC access code, or the Switched Loop LDN (main number). If no endpoint is specified, reorder is returned to the call.

- The following are administrable for a DID trunk group (maximum of 2 groups supported):

  a. DID trunk protocol (Immediate Dial, Wink, default=Wink)

  b. Dial Pulse (required) or Tone (desirable) (default=Dial Pulse)

  c. Expected Number of digits (1-4, default=3)

  d. Number of digits to delete (0 to 4 digits, default=0)

  e. Digits to add (a number from 0 (none) to 9999: add 0 to 4, default=0)

- The following are administrable on a trunk basis:
  a. Alpha Character label (7 characters max.)

  b. Trunk number (a unique flexible dial plan number, default in sequence with other trunks).

  c. Trunk Disconnect Timing (10 ms to 2550ms, default=500 ms)

DID numbers corresponding to pool dial-out codes (or facility access codes) can be used to avoid toll restriction, leading to toll abuse or fraud.

The options for each trunk group are as follows:

* DID Trunk Protocol
  * Wink start (default)
  * immediate start

* Dial Mode
  * pulse (default)
  * tone

* Expected Number of Digits
  * 3 (default)
  * 0 to 4

* Number of Digits to Add
  * 0 (default)
  * 0 to 4

CONSIDERATIONS IN USING DID TRUNKS

DID trunks are reliable and efficient, but they are more complex than loop-start or ground-start trunks. This is why it is especially important to be aware of the way the individual components of a DID system interact.

In particular, it is especially important to coordinate any MERLIN LEGEND system PBX installation and maintenance work at the customer's premises with the operating company. The PBX must be properly administered for DID service, and the proper type of DID service, before the operating company puts the trunk group into service. If DID trunks receive too many incoming calls while the PBX is "down" it is possible that the network will record enough integrity failures to remove the DID trunks from service. Callers to these trunks will receive reorder tone even after you're sure you've returned the PBX to service.

In order to avoid this possibility, some telephone companies offer a busy-switch which is located at the customer premises and connected to the CO. When testing is to be done, or the PBX powered down, the switch can be used to alert the CO, and in turn, the CO will ensure the DID lines are not removed from service.

Also, it is possible for the PBX to misinterpret the addressing information sent from the CO when the DID trunk is seized. This is called digit mutilation. When this occurs, the PBX receives only part of the addressing information. The LEGEND system will almost always recognize the mutilated digit sequence as incomplete addressing information and return reorder tone to the CO. It is statistically possible for digit mutilation to result in a valid, but incorrect DID code. If this happens the caller will get the wrong party.

And finally, DID service is efficient because it does not require a one-to-one correlation between a telephone number and a trunk. In a properly sized system this enables more incoming traffic to be routed over the DID trunks. However, if the system has been undersized (too few DID trunks to carry the traffic) it will appear that incoming callers are getting busy signals while DID users are sitting idle. The busy signals are actually reorder signals from the CO. The obvious solution to this problem is to install more trunks.

WHEN TO USE DID TRUNKS

DID trunks provide fast access to specific individuals. DID trunks should be considered when a customer has sales people who work with assigned territories, service people who work with 'assigned accounts, or any other organization seeking personal accountability. DID trunks, especially when combined with station features like bridged appearances, combine the flexibility of key system operation with the sophisticated options of the LEGEND system.

DID service is not available from all Central Offices, so always check availability before suggesting the service to a customer. Remember that DID trunks provide incoming service only. For outgoing calls use ground-start or WATS trunks.

TIE TRUNKS:
Simple Tie Trunk Operation

This application is perhaps the most common instance of the terms "trunk" and "line" being used interchangeably. This type of facility links two PBXs.

In typical operation a user on the local PBX would like to speak to a station user at a remote PBX over a tie trunk. The user on the local PBX lifts the handset to his telephone and gets dial tone from the local PBX. The user dials the access code for the tie trunk, the local PBX connects the telephone to the tie trunk, and the user on the local PBX hears dial tone from the remote PBX. The local station user then dials the extension number of the remote user.

The switching capabilities of *both* PBXs are utilized in a call placed over tie facilities, so this facility is properly called a "trunk."

\fBSimple Tie Trunk Operation\fR

Tie Trunk Call from
Sta. B to Sta. D

Tie Trunk

New York   Los Angeles
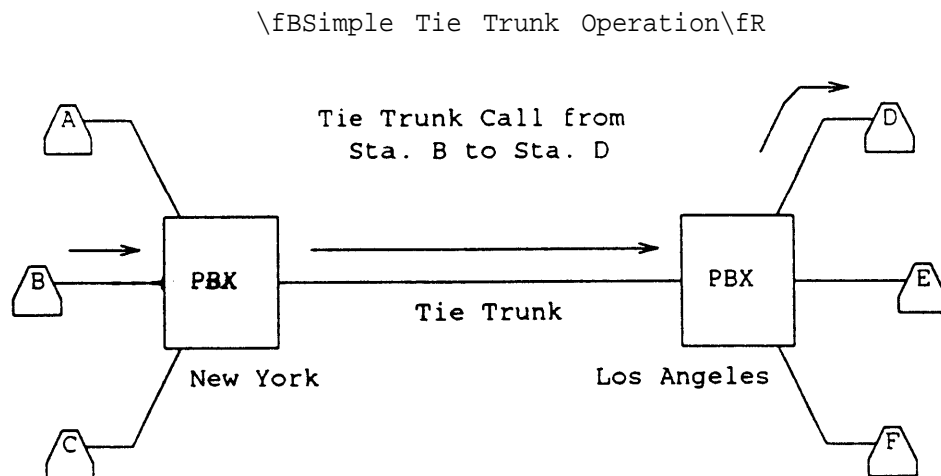
In special cases, tie trunks are used to join one PBX directly to another PBX. This may occur if multiple PBXs serve a single-customer at a single location. A typical example would be a corporate headquarters consisting of several buildings, each housing a separate division, co-located in a campus setting on a single premises. This is sometimes called a "back-to-back" connection.

Tandem Tie Trunk Operation

In the case cited above, a station user on one PBX used the tie trunk to reach a station user on a second PBX. This is basic tie trunk operation, and it is available on any LEGEND system.

A more advanced form of tie trunk operation is known as tandem trunking. This involves using a tie trunk from one PBX to reach a trunk or other network facility. not just a station user, at a second PBX. Using two or more facilities connected in series is called tandem operation.

In a tandem trunking operation a user on one PBX can access a facility that is not available on the local PBX, but is available on a remote PBX. For instance, consider the case of a company with offices in New York, Chicago, and Los Angeles. The New York office is linked with the Chicago office by a tie trunk, and the Chicago office is linked with the Los Angeles office by a tie trunk. A station user in New York needs to talk to an outside supplier in the Los Angeles area.

If tandem operation is supported at all of the PBXs, then the station user in New York can access the tie trunk to Chicago and, from Chicago, access the tie trunk to Los Angeles. When the PBX at Los Angeles has been accessed, the station user in New York can use a local, Los Angeles trunk to call the supplier. (There will be some transmission degradation in this type of connection unless a digital network is used.) Of course, these tie trunks can still be used to access other station users at the remote PBXs, just like ordinary tie trunks.

\fBTandem Tie Trunk Operation\fR

Sta. B (New York) uses Tie Trunk
to access a local Los Angeles Trunk
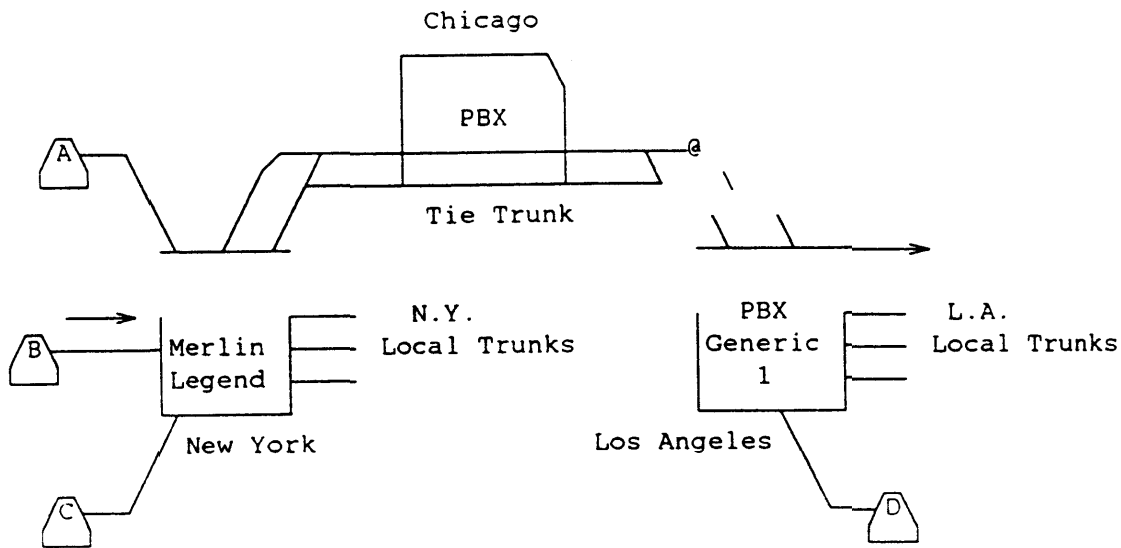
TIE LINES:
Transferring Calls

The MERLIN LEGEND CS supports trunk calls that come into the system and go out on another trunk without any operator assistance. However, the transmission code in the system does not permit multiple hops, and there is no automatic routing system which can be used for tandem trunking.

Never-the-less, with the LEGEND system it is very easy to transfer your calls over a tie line to another PBX. Whether the call you are on is a local line or a tie line call, the procedure is the same.

When you want to transfer a tie/local call begin by touching the Transfer button. This will give you dial tone.

Dial the tie line access code to connect to another switch. (For the sake of this example, let's say it's a Chicago switch.) When you hear dial tone, then dial the number of the extension in Chicago.

When the ringing tone is heard, complete the consultation transfer where you alert the called party that a transfer is being made by remaining on the line until they answer. Alternatively, you can complete the transfer by hanging up your station once the called station begins to ring; the transfer is completed automatically y. If the called party does not answer after you've gone on-hook, then Transfer Return will return the call to your station if the call is within your system.

If the user on the LEGEND system attempts to make another trunk-to-trunk transfer on the Chicago switch, or to dial a local Chicago outside number, this would constitute a second or multiple hop. It is this type of multiple hop the the LEGEND) system transmission code does not recognize, and the system does not support.
.

TIE TRUNKS:
Signaling and Implementation

The PBXs that terminate the tie trunk must be optioned to work in a compatible manner (for instance, to use tone or rotary dial pulse signaling), and the facilities must be engineered so they pass transparently through any central office.

The LEGEND system is administered for the appropriate tie trunk operation when the smite is installed. Tie trunks may use immediate start or wink-start operation, as described under DID trunks. Tie trunks may also provide automatic operation. With automatic operation no start dial signals are used. The seizure signal alone is sufficient to route the call to a predetermined destination. The call destination is determined when the trunk group is administered, and is usually the attendant.

In addition, a signaling format must be specified. Unlike loop-start, ground-start, and DID trunks, tie trunks may use special leads for signaling. These leads are called "E" (for "Ear") and "M" (for "Mouth"), and the system is referred to as "E&M lead signaling." The "M" (Mouth) lead sends signals from the switching equipment to the signaling equipment, and the "E" (Ear) lead listens for signals from the signaling equipment and carries them to the switching equipment. A signal from the local system to the distant system leaves on the "M" lead of the local system and arrives on the "E" lead of the remote system. If E&M leads are not used the tie trunk is referred to as being "Simplex."

In order to approach a.LEGEND system implementation with tie trunks it is really only necessary to know which signaling formats are supported and to determine which of these supported formats are required. The LEGEND system with the 400EM module supports Type 1 E&M Standard, Type 1 E&M Compatible, and Type 5 tie trunks.

The transmit and control signals for each signaling type are:

TYPE	TRANSMIT		RECEIVE	
	ON-HOOK	OFF-HOOK	ON-HOOK	OFF-HOOK
Type 1 Standard	ground	battery	open*/bat	ground
Type 1 Compatible	open*/bat	ground	ground	open*/bat
Type 5	open	ground	open	ground

* **Note:** An open circuit is preferred over battery voltage.

As its operation implies, the successful implementation of a tie trunk format will be dependent on matching the characteristics of both of the switching systems it connects. The preferred signaling formats for a tie trunk terminating in the MERLIN LEGEND system are:

TERMINATING SYSTEMS			NEAR END		FAR END	
FROM	TO	DESCRIPTION	MERLIN LEGEND SIGNALING MODE	Protected/ Unprotected	MERLIN LEGEND SIGNALING MODE	Protected/ Unprotected
LEGEND	S25/S75/Definity	Co-located	Simplex Type 5		Simplex Type 5	
LEGEND	S25/S75/Definity	Inter-office	Simplex Type 5		Simplex Type 5	
LEGEND	S85	Co-located	Simplex Type 5		Simplex Type 5	
LEGEND	S85	Inter-office	Simplex Type 5		Simplex Type 5	
LEGEND	Dimension	Co-located	E&M Type 1 Crept	Unprot	E&M Type 1 Std	
LEGEND	Dimension	Inter-office	Type 1 Crept	Prot	Type 1 Std	Prot
LEGEND	OTHER	Co-located	E&M Type 1 Crept	Unprot	E&M Type 1 Std	Unprot
LEGEND	OTHER	Inter-office	E&M Type 1 Crept	Prot	E&M Type 1 Std	**
LEGEND	LEGEND	Same site interbuilding	Simplex Type 5		Simplex Type 5	
LEGEND	MERLIN II	Same site interbuilding	Simplex Type 5		Simplex Type 5	

** **Note:** Plus a protection unit a little getting use to.

OFF PREMISES STATIONS

On occasion, customers need to locate telephones away from the principal area of the PBX or Key system terminal concentration. The telephone can use an off-premises or out-of-building arrangement. There is a major difference between the two types.

An off-premises telephone (OPT) is a single-line set that is located in another building and connected to the LEGEND system via a special arrangement with the CO.

> **The station has the same features as an on-premises single-line station except it is counted as an outside party in a conference call. Also, the message light feature will not operate with the OPT set.**

The OPT feature is sometimes used to provide service to executives at their homes. It allows them remote access to the **PBX** system features and services.

An out-of-building telephone, on the other hand, is connected directly to a system port even though it is not located in the same building. It may be an ATL or MLX set, single-line or multiline. Because it is direct, it has access to all system features. However, if the telephone is located less than 1000 feet from the switch, an IROB (in range, out of building) protector must be attached.

The OPT connection from a CO would be equivalent to a tie line; in effect, it is a dedicated line. Because it has the same electrical properties as a tie line, the 008 OPT module can be used to connect the LEGEND system to another LEGEND system, or another PBX or key system. The characteristics of the 008 OPT module are listed below:

It should be noted that an OPT connection to a CO requires that the LEGEND system be no further than 1 to **3** miles from the CO, depending upon the characteristics of the CO involved.

008 OPT: The Off-Premises Telephone 008 OPT Module is used to connect off-premises touch-tone telephones to the LEGEND system. The system software recognizes the 008 OPT module as a 012 module. Even- though the OPT module has 8 jacks, it uses 12 ports of capacity, thereby decreasing overall station capacity by four stations for every OPT module used.

Module	Station Type	Specifications
008 OPT	On-premises or off-premises single-line telephone	Capacity: 8 stations, 2 Touch-Tone receivers Notice to Telco: meets FCC 0L13C and :/FCC Class C Loop resistance: serves 2-wire loops to 1300 ohms, including stations Port losses: 3 db (both directions)
	Unit Load Rating	8.0
	max. distance supported	1-3 miles between Legend and CO

When an Off-Premises telephone is connected to the LEGEND system via the 008 OPT module certain accessories must be used for grounding and protecting the system from power surges, electromagnetic interferences, and electrostatic discharges. These components are specified in Section 5 of the LEGEND System Reference Manual.

Service technicians should be aware that the 008-OPT module is the same as the 008-DID module except that the battery strap is in the On position for OPS, and in the Off position for DID. The factory presets the strap and labels the module accordingly, based on the DOSS order.

The OPT trunk from the CO provides -48VDC on the tip/ring interface to the OPT station.

The MERLIN Off-premises Telephone lnterface (OPTI) (PEC 2302-OPI) cannot be used with the LEGEND system.

T-1 (DS1) SERVICE

OVERVIEW AND OPERATIONS

The goal of the T-1 (DS1) section is to present basic information about T- 1 networking, what it is, how it operates, and what it means to the customer. The section does not explain how the MERLIN LEGEND Communications System interfaces with ISDN-PRI.

Further, this Note does not cover the indepth details of signaling and encoding on which T-1 service is based. Instead the focus is on giving an overview of how T-1 fits into the telecommunications needs of customers.

T-1 signifies a virtual revolution in how analog and digital signals are sent over a network.
It provides a two-way connection at 1,544,000 bits per second. This can be stated as 1.544 Megabits per second (Mbs), which is also called "digital signal level one" (DS1). Generally the designation "T-1" means any transmission line or connection running at 1.544 Mbit/s. In a stricter sense, T-1 is applied to the system of copper wire cables and amplifiers or regenerators that reinforce the digital signal at intervals of approximately one mile.

AT&T, telephone companies, and other interconnect companies all provide T-1 service. Also, a very large customer may own its own T-1 system. A system can be made up of copper wire cables, microwave. optical fiber, or other media. It is not unusual for three or more carriers using various media to be involved in providing T- 1 service to a customer. Often it can be as many as six or eight carriers.

The standard T-1 line consists of 24 channels. One T-1 line replaces 24 of the voice grade analog copper wire pairs known as 3002 lines. One T-1 can also transport 150 data channels; each would need a full voice grade line on the analog network. To be most economical, T-1 transmission can combine voice and data signals   simultaneously.

If a customer connects to AT&T, the customer premises PBX must be compatible with a 4ESS (Generic 13 or higher) toll office switch. If the connection is to the telephone company, the PBX must be compatible with the telco 5ESS central office switch. (The MERLIN LEGEND CS is certified for use with the AT&T 4ESS switch and the 5ESS switch for T-1 service.) It is possible to get fractional T-1 (FT1) service from most T-1 providers where the user can designate time slots for sending and receiving by using only part of the 24 circuits available in a T-1 link. The availability of FT1 will be of special interest to many GBS customers. If customers have six or more circuits of any kind, fractional T-1 is likely to be more economical.

To understand better the use of T-1 it's necessary to appreciate the advantages that T-1 offers to the communications system operator. These may be headed under the broad categories of Network Control, Reliability, and Quality.

When telecommunications was solely dependent upon the use of voice grade 3002 lines, it was necessary to know weeks (or even months) in advance when a user would require a change in the way network lines were being used. It usually took that long for the telephone company to make the rearrangement. With T-1 service one of the major Network Controls the user gains is the quick ability to configure the system within minutes. The network manager reassigns the channel resources where they are needed to meet changing daily business requirements, emergencies, or to improve productivity.

Another Network control that is of vital user interest is diagnostic control. With the system split between half a dozen carriers, it is necessary to be able to monitor the various parts on a continuing basis. This means the user has the ability to locate faults and quickly get the proper maintenance group to restore

service, quickly and efficiently.

In the past it was necessary to establish a new circuit for virtually every application a customer wished to use. Whether it was teleconferencing, facsimile, hifi audio, video, or computer transmissions, generally a new line or trunk had to be run. It required a lot of time (perhaps months) and money to install each circuit. With T-1 all these circuits can be simplified into a single T-1 circuit.

DESIGN, SIGNALING & IMPLEMENTATION

Design of a T-1 network can be focused on the public network, a private network (user owned), or a hybrid combining advantages of both. Regardless of which is used, the MERLIN LEGEND Communications System is ideal for connecting the customer to the T-1 service. It's 100 DS1 Module allows connecting one 24-channel line to the PBX at a cost far less than that of connecting 24 voice grade lines to individual ports. .

In addition to the T-1 channels. the network requires the use of regenerators and multiplexer to boost the signals and combine the information of the channels so it can be directed into a smaller number of channels. The repeaters ensure high quality of the transmissions. The multiplexer allow more efficient and productive use of the channels.

An advantage of T-1 is that it can send both voice and data over the same circuit.* This means the voice signal must be converted to digital. A number of coding techniques are used to convert and compress the analog voice signals into digital signals. AT&T uses a method called Pulse Code Modulation (PCM). These encoding methods are what make it possible to transmit the traffic of 24 voice grade lines over one T-1 line.

Before information can be sent over the channel it must be put into the proper format or frame. The minimum need is that very 193rd bit be used to form a D4 or Extended Super Frame (ESF) pattern. This minimum formatting will allow the customer to connect to any T-1 terminal device and route DS-1 circuits through any carrier facility.

Further examination of the signaling and implementation required for T-1 circuits would warrant an indepth technical presentation. The goal here is to show that T-1 can be of value to GBS customers.

Note, however, that T-1 service should not be the only dial tone available to a LEGEND system. It the T-1 goes out of service the entire system would go down. This means that every T-1 system should also be connected by ground-start trunks to the CO.

Potential Problems/Limitations

There are a number of factors, however, which could impede the implementation or T-1 service and should be considered. They include:

- The question of standards for all T-1 components is still unresolved. This means many manufacturers of components are using proprietary interfaces and encoding methods. In turn this means the customer is often forced to use the same equipment at both ends of a T-1 network.

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4. MERLIN LEGEND Communication System requires ISDN-PRI to support digital data via the network.

The technology for T-1 is moving rapidly. The encoding system and components are not always upgradable to the newer innovations.

- If the customer application requires maximum network reliability, automatic dialing back-up in case of digital facility failure could prove costly.

Do GBS Customers Need T-1?

Consider this: a standard T-1 circuit consists of 24 analog voice circuits. The cost of 24 long distance circuits is more expensive than a T-1. Within the same Central Office a T-1 line is less expensive than the use of 5-to-7 voice grade lines. This may also be true where only 2 voice grade lines are involved, depending on the local traffic. Where fractional T-1 (FT1) is available, the cost break point is even more favorable.

In fact, some GBS sized customers are losing money by not converting to T-1 service.

With a MERLIN LEGEND CS, a customer can use the T-1 lines as Loop-start, Ground-start or Tie (4-wire E&M lines). This means the lines can be programmed to emulate, or look like, these facilities. (DID is also available on T-1 service, but this is not supported on the LEGEND system.) When the 100D module is programmed for T-1, it can be set to emulate which ever type of tine is needed (GS/LS/Tie - or OPX). The programming must, however, match the settings for a given line assigned to it by the Central Office. Complete details for administering the correct T-1 settings can be found in Section 4 (page 4-44), of the LEGEND system Installation, Programming, and Maintenance Manual (555-410-140).

In order to use T-1 services the LEGEND system is connected to the network by a data communications device called a Channel Service Unit (CSU). The next section explains the purpose of the CSU and the two types available. It also covers the use of multiplexers used to increase the efficiency of the T-1 channels.

DATA COMMUNICATIONS EQUIPMENT

This section is an overview of various equipment which is used to enhance the capability of data communications and or Integrated Services Digital Network- Primary Rate Interface (ISDN-PRI) operations. The Channel Service Unit is necessary to connect the MERLIN LEGEND system to the Digital Signal One (DS1) or PRI facility. The Multiplexer is designed for use on large systems where it is necessary to combine the B channel input into a single digital stream for significant cost savings.

A hypothetical customer has installed a LEGEND system and is subscribing to MEGACOM service to handle voice traffic. The question is, Which CSU should you order?

The LEGEND system will operate properly on either of the AT&T CSU's identified below. They both can transmit up to 1.544 Mbps. While the Model 551-T1L1 is the lowest price unit, the ESF-T1 model provides a more extensive range of features and diagnostic tools. The ESF-T1 should be the unit of choice if the customer is capable of performing diagnostics on the circuits, and if the network smite will also be used to transmit data. The 551 model should be used if the transmission speed is 56Kbps and clear channels are not a primary concern.

CHANNEL SERVICE UNIT (CSU)

The Channel Service Unit (CSU) provides the interface between the 100D module and the DS1 facilities. Two AT&T models are available for use on the Legend system: the ESF T1 CSU and the 551 T1 L1 CSU.

- Model 551-T1L1
  Reg. Number G1 47226819-/DE-N

  The 551 T1 CSU is a full-duplex modem, provides data terminal equipment (DTE) with access to synchronous DS1, 1.544-Mbps lines. The CSU accepts data from the DTE and transmits it to the serving central office. It also receives signals from the serving CO and transmits them to the DTE. The CSU has three primary uses:

  - To terminate a DS1 transmission system on a PBX

  - To ensure that signals entering the public network from the DTE comply with the transmission system's requirements as defined in AT&T Technical Reference 62411 and FCC Part 68 Requirements.

  - To provide maintenance, diagnostic. and testing capabilities.

  Model ESF-T1 Reg. Number GIC 47216544-DE-N

  The ESF T1 CSU harnesses the real-time diagnostic capabilities inherent in facilities using the Extended Superframe format (ESF), such as AT&T ACCUNET T1.5 Service. This highly productive diagnostic tool evaluates circuit performance and records performance data, without service interruptions. The ESF T1 transcodes DID through D4 formatted signals into Extended Superframe format for transmission at 1.544 Mbps over T1 lines.

  The Facility interface Codes for the CSUs are:

  - Digital D4 Framing 4DU9B

  - Digital ESF Framing 4DU9C

  - Digital ESF and B8ZS 4DU95

  Note: If a non-AT&T CSU is used on a Legend system, AT&T will not accept the responsibility for its installation, connection, or testing.

The ESF T1 CSU can connect the DS1 network by using the D4 or extended superframe format (ESF). Also, it is the only CSU to provide the B8ZS line coding needed to transmit a 64-Kbps clear channel.

The ESF T1 CSU can be mounted in a relay rack or on a shelf as a stand-alone unit. Power can be provided by plugging the CSU into a 117-VAC outlet.

Installation of the ESF T1 CSU involves setting the switch options. mounting the CSU as needed, and connecting the wires from the l00D module and the DS1 network. The procedures for installing the CSU can be found in Section 3 of the Legend Installation, programming and Maintenance Manual.

MULTIPLEXER (MUX)

Multiplexer are devices that combine several individual information-carrying channels for transmission over an aggregate link. This is done by allotting this aggregate link to multiple users, in turn to constitute different intermittent channels or time slots (time division multiplexing for digital transport.)

A Multiplexer in general would only be required for large PBX systems using multiple 100D DS1 Modules where all B channels are assigned to various ISDN services. Customers would have complex voice/data networks, have multiple locations, and usually a large investment in private line facilities.

AT&T ACCULINK Multiplexers, for example, can combine up to 128 input channels into a single digital stream for cost-effective protocol-independent transmission. Each data ACCULINK channel can be programmed individually for a wide variety of data rates, ranging from 300 bps to 64 Kbps. Special application multiplexer, such as the ACCULINK 740 and 741, are designed to drop/insert data or video channels from/to a DS1 data stream originating from a digital PBX such as System 25 or System 75/Generic while passing the remaining DS1 voice channels through to the digital PBX.

A special type of multiplexer which can be used on a MERLIN LEGEND system is an ACCULINK Multiplexer. It would connect on the PRI side of the switch to the 100D DS1 Module. A 740/741 mux provides preselected voice, data. or video signals to be added to or removed from the DS1. while allowing the remaining DS1 voice channel to pass to the DS1 module. By the use of optional channel cards, a 740/741 can support up to 19.2 kbps asynchronous, or 56/64 kbps synchronous transmission.

The following table identities the AT&T DS1 services available to customers.

DS1 FACILITY SERVICES

Digital Signal 1 Service	Description	ISDN	T-1
Megacom (Megacom WATS) Service	An outgoing domestic long-distance service used in place of WATS service.	x	x
Megacom 800 Service	An incoming, domestic toil-free, number service for voice calls	x	x
Megacom/Megacom 800 Service	Adding Shared Access for Switched Service (SASS) allows Megacom and Megacom 800 service on the same trunks.	x	x
Megacom 800 Service with Dialed Number Identification Service (DNIS)	An incoming, domestic toll-free number service that provides voice information service on an interactive basis. Calls can be routed to separate departments or prerecorded messages can be played for different groups of callers.		x
Multiquest Service	An incoming domestic 900 number service for voice and data calls.	x	x

Digital Signal 1 Service	Description	ISDN	T-1
Multiquest Service With DNIS	An incoming 900 number that provides callers with voice and data information service on an interactive basis.		x
software Defined Network (SDN) Service	A virtual private networking service for voice and circuit switched data calls (Up to 56 Kbps). SDN lets businesses use portions of the AT&T Switched Network in concert with their dedicated private line networks. The system, however, does not support "uniform dialing plan," which is necessary for complete integration with SDN.	x	x
ACCUNET Switched Digital Service	A digital switching service between subscriber data stations and far-end connection. Useful for batch data or file transfer, high-speed faxes, etc.	x	
Digital Data Transmission	LEGEND supports it in ISDN-PRI. LEGEND does not support it in straight T-1.	x	

Digital Signal 1 Service	Description	ISDN	T-1
Station Number Identification/ Automatic Number Identification (SID/ANI) Service	A caller identification service for systems with display telephones, call report systems, etc. ** SID allows the called station to display the station number of the caller. ** ANI allows the called station to display the billing number (main telephone number) of the caller. subscribers can choose to send their own calling information to other subscribers for display or suppress the out- going caller identification. Note: The availability of the caller identification information may be limited by local conditions.	x	

SPECIAL HOTEL/MOTEL TRUNKS

OPERATION OVERVIEW

In certain areas of the country there are special trunks for hotels and motels about which you should be aware. At one time they were widely used but today they are found generally in rural areas. The trunks generally are called HOBIC lines by telephone company personnel. The acronym stands for Hotel Billing Information Center.

Actually, there is nothing different about the lines; it's the service that's different. The service is the Hotel Billing Information System (HOBIS), which provides automatic time-and-charge information to the inn each time a guest makes a call.

A similar service known as Centralized Charge Quotation Service (CCQS) provides the same information for hospitals, law firms, advertising agencies, and other customers who need to keep accurate records.

A hotel/motel subscriber to the telephone company's HOBIS service would be equipped with the trunks and a special printer. A guest initiates an outside call over a HOBIC trunk by dialing a special entry code, most often the number "8". The HOBIS system immediately begins computing time and charges and relays the information to the HOBIC center. When the call is completed, the center automatically sends a signal over a data line to a modem which is connected to the printer. The printer lists the time and charges for the call. This saves the motel operator or desk clerk from having to request the time/charge information for each call. It also saves the hotel/motel dollars by allowing it to add the charges to the guests' room bill before they checkout.

The AT&T Horizon PBX offered a special Hotel/Motel software feature package which used Hobic lines. The feature package provided the software, modem, printer, and RJ11 line connector. Recently, sophisticated optional accounting systems, such as CAS PLUS Hospitality Version available on the MERLIN LEGEND Communications System have replaced the Hobic method, except in smaller, family-owned "mom-and-pop" type motels. These new systems do not use Hobic at all; they accumulate SMDR data and calculate the price of each call based on rate tables included with the Call Accounting software.

MAINTENANCE AND TROUBLESHOOTING

Whenever installing or programming lines/trunks there can be problems encountered. This is a quick reference guide to use when difficulty occurs.

> When difficulties arise on the lines/trunks, consult the Maintenance section (page 5-39), LEGEND Installation, Programming, and Maintenance Manual (IPM).
> Also consult the Lines/Trunks Error codes/Solutions matrix in the Troubleshooting section (page 5-46), in the IPM.

HARDWARE

The Legend system hardware most directly connected with Lines/Trunks are the Control Unit Modules. The specifications for each module are presented later in this section. Also the system uses different types of lines/trunks for the different individual functions of each operating mode. In PBX mode the system can use the following:

- Loop-Start (LS) trunks
- Ground-Start (GS) trunks
- Tie trunks
- Direct Inward Dialing (DID) trunks
- ADS 1 facility programmed for either T1 or ISDN-PRI operation.
- Centrex Service lines - Loop Start*

A Key system can use:

- Loop-Start lines
- Tie Trunks
- A DS1 facility programmed for either T1 or ISDN-PRI operation
- Centrex Service lines - Loop Start
- Ground start when registered as MF FCC classification.

A Behind Switch system can use:

- Loop-Start lines
- Tie Trunks
- Centrex Service lines - Loop Start
- A Ground-Start line when registered under the MF FCC classification.
- Direct Inward Dialing lines

The LEGEND system with a basic carrier has five slots for these modules. Up to two expansion carriers can be added, each one adding six slots, for an overall system total of 17 slots.

The system supports 13 types of line/trunk and station modules. The table on the next three pages identifies the various modules and their uses.

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4. For dial-tone only in PBX mode. Switchhook flash or pooled Centrex lines will not work to activate Centrex features.

LEGEND SYSTEM CONTROL UNIT MODULES			
MODULE	LINE/TRUNK TYPE	STATION TYPE	SPECIFICATIONS
008	n/a	Analog multiline telephone; Call Management System; analog data with a general purpose adapter	Capacity: 8 analog stations Signaling: analog multiline telephone protocol (40 kbps) Loop range: 1000 ft in-building or in-range out-of-building with analog IROB protectors) service only
008 MLX	n/a	MLX telephone; digital data service (the ISDN 7500B Data Module)	Capacity: 8 digital stations, each with 1 or 2 endpoints. including the following station types: *digital voice only *digital voice with Voice Announce to Busy *digital voice and digital data (via ISDN 7500B Data Module) *digital voice and MFM *digital data only (ISDN 7500B Data Module) Signaling: ISDN-BRI S/T protocol (2 64-kbps B channels, one 16-kbps D channel), on a passable bus. Power: 48-VDC phantom power to telephone: 48 VDC over a separate pair (7-8) to an operator DSS console Loop range: 1000 ft. in-building and in-range out-of-building (with MLX IROB protectors) service only
008 OPT	n/a	on-premises or off-premises single-line telephone	Capacity 8 T/R stations on 2-way voice transmission path with support for telephones with message-waiting LEDs: 2 TTRs Ringing current: 105-Vrms. 30-HZ sinusoidal ringing super-imposed cm -48 VDC; a ring generator must be installed in the power supply module of each carrier that has a 008 OPT module. REN: <1.0 per port Disconnect signal: 900 ms (T/R short for answering machines, G3 Fax, etc.) Switchhook flash detection 300-1200 ms Loop Resistance: Serves 2-wire loops to 1300 ohms, including stations Notice to Telco: Meets FCC Class c

MODULE	LINE/TRUNK TYPE	STATION TYPE	SPECIFICATIONS
012 (T/R)	n/a	Single-line telephone; MERLIN attendant; MERLIN MAIL Voice Messaging System T/R adjunct (such as an answering machine or fax machine); analog data device (such as a modem)	Capacity: 12 T/R stations on 2-way voice transmission path with support for telephones with message-waiting LEDs. 2 TTRs Power: 21-VDC, 600-ohm battery source Ringing current: 105-Vrms, 30-Hz sinusoidal ringing superimposed on -48 VDC; a ring generator must be installed in the power supply module of each carrier that has a 012 module REN: <1.0 per port Disconnect signal: 900 ms (T/R short for answering machines, G3 Fax etc) Switchhook flash detection: 300-1200 ms
100D DSI	T-1 or PRI	n/a	Capacity: 24 lines/trunks for voice and analog data or 23 lines/trunks for voice and data with 1 channel used for signaling. Mode: multiplexes 24 or 23 line/trunks into 1 facility and demultiplexes 1 facility into 23 or 24 lines/trunks speed: Up to 64 kbps Signalling: DS1 over 4-wire; T-1 uses robbed-bit or common channel; PRI has ISDN-PRI (23B + 1 D)
400/TTR*	Loop-start and TTR	Power failure transfer (PFT) telephone	Capacity: 4 lines/trunks. 4 TTRs. 1 PFT telephone Signaling: Loop-start
400 E&M	Tie trunk	n/a	Capacity: 4 tie lines Method of completion: automatic start, immediate-start. wink-start, or delay-did-start Signaling: E&M type IS. type 1C. type 5
400 GS/LS/TTR	Loop-start or ground-start and TTR	PFT telephone; button needed for ground-sun PFT telephone	Capacity: 4 lines/trunks, 4 TTRs, 1 PFT telephone signaling: Loop-start or ground-start, optioned per port

MODULE	LINE/TRUNK TYPE	STATION TYPE	SPECIFICATIONS
408*	L00p-start	Analog multiline telephone; Call Management System; PFT telephone	Capacity: 4 lines/trunks, 8 stations. IPFT telephone Station signaling: analog multiline telephone (40 kbps) Line/trunk signaling: Loop-start line/ trunk: analog voice Loop range: 1000 ft.. in-building and in-range out-of-building (with analog IROB protectors) servicc only
408 GS/LS	Loop-start or ground-start	Analog multiline telephone; Call Managemcnt System: PFT telephone	Capacity: 4 lines/trunks, 8 stations. 1 PFT telephone Station signaling: analog multilinc telephone (40 kbps) Line/trunk signaling: Loop-start line/ trunk (optional per port): voice Loop range: 1000 ft. in-building and in-range out-of-building (with analog IROB protectors) service only
800*	Loop-start	PFT telephone	Capacity: 8 lines/trunks, 2 PFT telephones Signaling:R loop-start
800 DID	Direct Inward Dialing	n/a	Capacity: 8 trunks. 2 TTRs Transmission: incoming calls only; 2-way (l-pair) fixed impedance to DID trunks; no outgoing calls Signaling loop-reverse battery; wink-start or immediate start; accepts touch-tone dialing
800 GS/LS	Loop-start or ground-start	PFT telephone; button needed for Ground-start PFR . .	Capacity: 8 lines/trunks. 2 PFT telephones Signaling Loop-start or ground-start

* Note: Although these MERLIN II module are supported in the MERLIN LEGEND Communications System. the 400 GS/LS. 408 GS/LS. and the 800 GS/LS are the recommended modules for use in the LEGEND system.

This section lists the LEGEND system trunk and off-premises modules and the appropriate PEC and Comcodes. More complete code information can be found in Section A of the Systems Manual (555 -610-200).

Line/Station Modules	PEC	Comcode
100D DS1 module	61491	105461560
400 E&M module	8303-200	105311401
400 GS/LS/TTR module	61483	105627988
400 (TTR)	61379	105408892
408 LS ATL module	61482	105512495
408 GS/LS ATL module	61481	106064678
800 DID module	61488	105628002
800 GS/LS module	61484	105627996
800 LS module	61384	105351100
008 OPT module	61489	106387525
008 ATL module	61485	105351092
008 MLX module	61486	105628010
012 Basic Tel	61487	106397631

Reusable Modules from Merlin II (R3)

The following modules used in Merlin II Release 3 system can also be used in the Legend system.

- 391A Power supply module
  (39lA1 recommended; found in later MERLIN II R3s)

- 800 Line/trunk module

- 400 Line/trunk module

- 400 E&M line/trunk module

- 012 Basic telephone module

- 008 Analog station module

- 408 Analog line/trunk & station module

ADMINISTRATION

To properly administer a MERLIN LEGEND system at the time of initialization requires that care be taken during the planning phase of the sales activity. This is especially true if the system is to connect to a number of different trunks and lines, such as DID, Tie, or ISDN. in addition to using local CO Loop-start and Ground-start lines. To assist in the planning of the system, sample, filled-in copies of the planning/administration forms are available. They can be found in the LEGEND PBX System Planning Manual (555-610-113), and the Key System Planning Manual (555-610- 112). These forms should be reviewed before starting to complete an actual set of forms for customers' systems.

Also, keep in mind, that any MERLIN LEGEND CS configuration that requires the use of T-1 and/or PRI trunks, must be reviewed by the NSAC. The Planning Manuals contain directions for sending the appropriate forms to the NSAC.

CONSIDERATIONS

Special care must be taken when planning and ordering trunks and lines for the MERLIN LEGEND CS. The most important aspect which the planner must keep in mind is that it is critical to investigate what services the local interconnect company can provide. For example, if a Behind Switch system is to be connected to a telephone company Centrex switch, the Central Office must be able to provide Loop Start lines capable of sending a disconnect signal. Otherwise the Legend system will ultimately hangup as more and more trunks are considered in use because no disconnect signal was received by the system.

Also, keep in mind that the use of Ground Start Centrex lines are not supported by the LEGEND system. So ensure that the CO does not include any GS lines in the block of Centrex lines it assigns to the Legend system.

When consulting with the Central Office regarding Lines and Trunks, also determine the proper timer setting for the disconnect signal from the CO.

When ISDN applications are being considered, the networking should be through an AT&T 4ESS switch (Generic 13 or higher), operated by Network Systems in order for the customer to obtain the largest selection of ISDN services. However, the MERLIN LEGEND Communications System also will operate properly when connected to a telephone company 5ESS switch.

REFERENCES

GBS SCAN Competitive Newsletter, "Everything You Need to Know About ISDN", Nov/Dec 1988.

MERLIN LEGEND Key System Planning Manual (555-610-112)

MERLIN LEGEND PBX System Planning Manual (555-610-113)

MERLIN LEGEND Data Guide (555-610- 114), Chapter 2

MERLIN LEGEND System Reference Manual (555-610-110)

MERLIN LEGEND Installation, Programming, and Maintenance Manual (555-610-140)

AT&T ESF T-1 Channel Service Unit User's Manual (999-100-305), 1988.

Lescinsky, F.W.; FDS ISDN PRI Digital Data, Iss. 1.1; 8/13/91

Wright, A.E.; FDS ISDN PRI Service, Iss. 1.0; 2/13/90

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