Basic and Primary Rate
ISDN

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Preface

ISDN does not provide new services in as much as it makes it possible to combine multiple capabilities onto a single interface. In the past if a user needed Switched 56, X.25, and 2-wire voice services they had to order a separate circuit for each service. ISDN allows all of these services, and more, to be accessed from the same circuit. Each time a connection is needed the required service is requested by the user's equipment. Given the proper terminal equipment at the customer premise one ISDN basic rate service can connect to a variety of services and equipment. This includes 9.6/56/64 Kbps X.25 psdn access, 1.2 to 64 Kbps data (ISDN at both ends), 128 Kbps data (ISDN at both ends), V.32/34/90 analog modem connections and conventional telephone services. There are of course limits on how many of these connections can be active at any one time.

Built around a set of ITU standards ISDN comes in two major varieties, Primary rate and Basic Rate. While the Signaling protocols are very similar between Basic Rate and Primary Rate ISDN the physical interfaces that carry the signaling and user data are very different. Primary rate has many similarities to T-1 or E1 such as the line rate and the same 4-wire transmission medium. Basic Rate uses a physical layer unique to the ISDN world.

In the US and Canada ISDN signaling is defined by a set of standards from Bellcore called National ISDN. Throughout the rest of the world the most common standard is called NET3 or ETSI. Both the NET3 and National ISDN standards have their roots in Q.921 / Q.931. This document will refer to the Q.921 & Q.931 standards. An attempt has been made to explain some of the differences between National ISDN and NET3, however, some of these differences may have been overlooked.

This document contains frequent references to AT&T, Nortel, Siemens, Bellcore, The National ISDN Users Forum (NIUF), the ITU among others. These companies and organizations are very important to ISDN on a national and international basis. If any representation of their operation, function, or equipment description is in error I apologize and would be happy to correct problems that anyone finds with this document.

Contents

Introduction

ISDN is based around two types of channels. The first is the **Bearer** or **''B''** channel. B channels are 64 Kbps clear channel on which the user information is placed. This information can take several forms such as data or PCM encoded voice. These may be connected, via a circuit switched connection, to the remote terminal much like a phone call or dial up modem connection.

The second type of channel is the **Common Signaling** or **''D''** channel. It is the function of the D channel to carry all information needed by the Central Office switch and the user terminal to establish connections, terminate connections, report status, etc.. As an example when a router invokes dial backup via a built in ISDN interface it will request a circuit switched connection by sending a message to the local telephone switch over the D channel. Once the switches have established the connection to the answering end the user data flows transparently over the B channel. To disconnect the call a different message is sent over the D channel. Once the B channel has been disconnected it is ready for the next connection. Typical connection times for call setup are 3 seconds or less for ISDN circuits.

The most common ISDN service is called basic rate or **BRI**. This service provides two B channels and one D channel. The D channel operates at 16Kbps. Basic rate is referred to as **2B+D** and is intended to be the access point for customer equipment such as ISDN telephones, ISDN Fax and other ISDN terminal equipment.

For higher capacity ISDN is also available in primary rate or **PRI**. This service differs depending upon the country where it is deployed. In North America, Japan and Korea PRI supports 23B channels and a single D channel. In the rest of the world it provides 30B channels and one D channel. These are referred to as **23B+D** and **30B+D** respectively
An understanding of ISDN signaling requires an examination of its use of the first three layers or ''chained layers'' of the seven layer OSI protocol stack. Functionally these three layers provide the standards that manufacturers use to guarantee compatibility between the host, or end user equipment, and the node, or telephone switching equipment.

Layer one is the physical layer. In ISDN this includes the physical and electrical characteristics of the interface necessary to connect the ISDN terminal equipment to the circuit. Another example of a layer one standard would be the RS-232-C standard. In the U.S. there are two different Layer One BRI standards defined by ANSI T1.601 1992 and ANSI T1.605 1991. Layer one defines transport of all customer and signaling data between the customer premise and the local telephone CO.

Layer two is the data link layer. ISDN layer two is called Link Access Procedures D Channel or LAPD. Only the D channel uses this layer. Based on the HDLC model LAPD provides a method of error free communication over the physical layer. SDLC and LAP-B (X.25) are examples of other data link layers. ITU Q.921 defines layer two standards for LAPD.

Layer three is the network layer. Included in layer three is the actual information, needed by customer premise equipment and telco switches, to setup and terminate a connection. Layer three also allows needed information to pass between the ISDN terminal equipment and the CO. Layer three information is put into a layer two frame and transmitted over layer one. ITU Q.931 defines standards for layer 3.

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ISDN Connection Points and Equipment Types

Figure 1a shows the ITU ISDN reference model with equipment types and interface points. Figure 1b shows three of the most common implementations of BRI for data applications and their relationship to the ITU ISDN reference model. Figure 1c and 1d show the typical PRI installations.

Line Termination or **LT** equipment is the point at the telco switch where the basic rate circuit is terminated.

Network Termination or **NT** equipment is the customer premise termination for the basic rate circuit. Among its functions it must format the signal for transmission, respond to loop commands and report status. The NT can be compared to a CSU or NTU in some respects.

NT2 equipment is defined as concentrating and multiplexing equipment. This could be a PBX or terminal controller. In many applications the NT2 is unnecessary. It should be noted that even though the S and T reference points have different locations in figure 1a. 1 they are identical with respect to operation and ISDN devices with this interface are usually referred to as S/T interface devices.

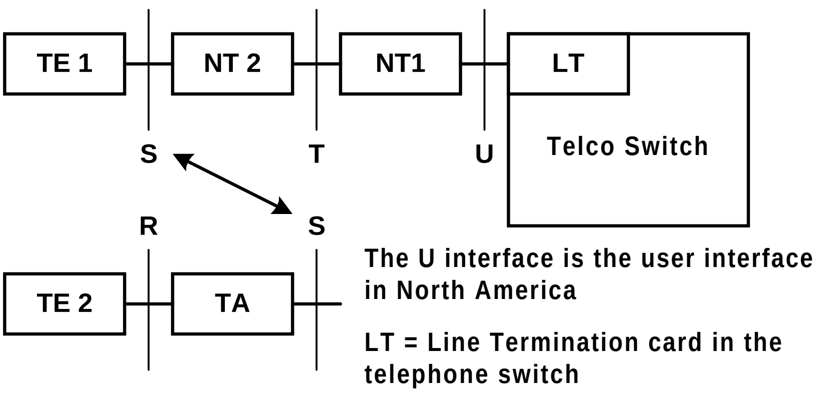
ISDN terminal equipment, or **TE1**, is equipment that is designed to connect directly to the S/T interface. This includes ISDN telephones or a PC with one of the commercially available internal ISDN cards.

An ISDN terminal adapter or **TA** is used to connect a non-ISDN terminal to the S/T or U interface. It should be noted that the R reference point is not defined by ITU but in most situations would be RS-232, V.35, Ethernet, etc.

Non-ISDN Terminal equipment is referred to as **TE2** equipment. This includes a DTE that needs to be connected to the ISDN Basic Rate circuit but does not have an S/T or U interface built in. Anything presently connected via RS-232 or V.35 to a modem, such as a PC, is to be considered TE2 equipment and the non-ISDN interface is referred to at the **R** reference point.

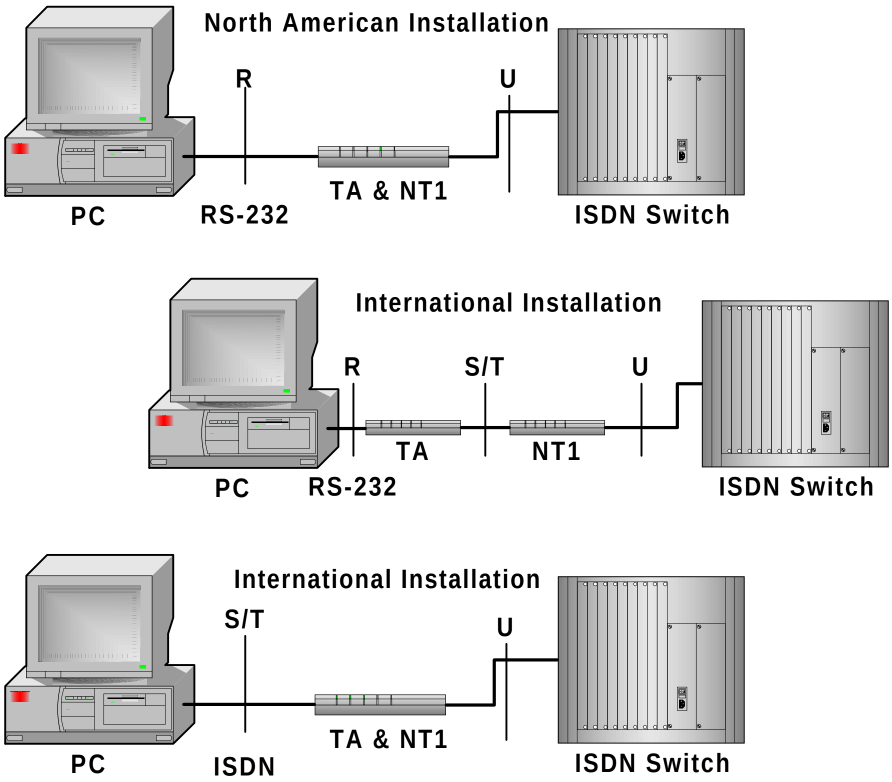
The **S/T** interface is a four-wire bus between the NT1 and the ISDN terminal equipment. S/T interfaces are defined in ANSI T1.605 1991 and ITU I.430 layer 1 specifications.

The **U** interface is the two-wire point to point interface between the LT and the NT. In the U.S. it will often be found that the NT is not a standalone piece of equipment. When this is the case the unit will be referred to as a U interface Terminal Adapter. The U interface is the customer connection point in North America while the rest of the world uses the S/T point.



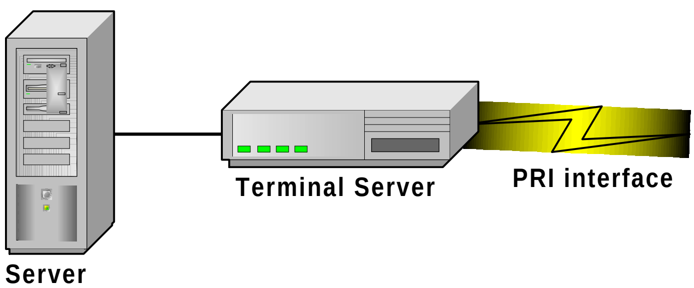
ITU BRI Reference Points and Equipment Types
Figure 1a

Both the U and S/T connection points are considered to be layer 1. A standalone NT, while it converts the format from U to S/T, has no D or B channel *intelligence;* that is, it does not examine the data on the D or B channels at any higher level than at the one or zero level. Figure 1b shows BRI implementations in real world situations.



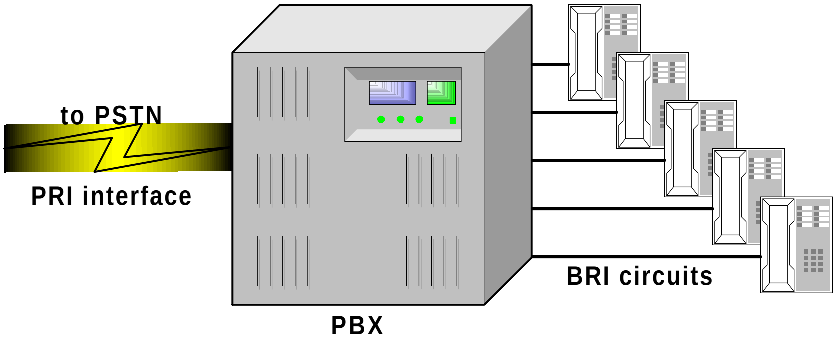
Reference Points in Typical Installations
Figure 1b

The real world implementation of PRI is less complex than BRI. This is because PRI does not support multipoint connections. Generally the reference model can be reduced to the NT and TE. Shown below are two implementations of PRI. Figure 1c shows a simple remote access implementation using PRI for dial up access service to a LAN.



Typical terminal server Installations
Figure 1c

Figure 1d shows an implementation typical in PBX configuration. This allows a single PRI interface to support multiple telephones for connection to the PSTN. In this configuration there could be telephone sets in excess of the 24 telephones because it is unlikely that all of the users will be on the phone at the same time.

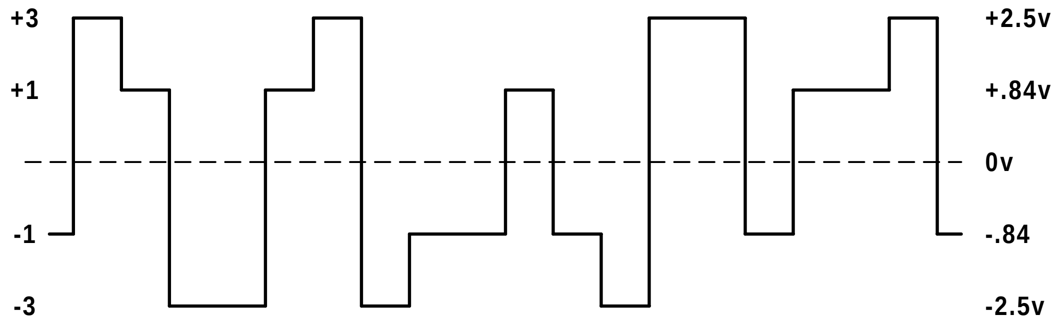


Typical PBX Installations
Figure 1d

One of the advantages to using PRI is its ability to work with Computer Telephony Integration or CTI. ISDN is capable of delivering the calling party number for each incoming call. This information can be interfaced a database and a call distribution application. In call center applications a desktop PC can be used to provide caller information to the call center operator.

BRI Layer 1 "U" Interface

The U reference point is a 2 wire full duplex connection to the telco CO. To transmit this signal in both directions simultaneously a method called ‘’ECHO CANCELER WITH HYBRID’’ is employed. In this scheme the NT learns it’s echoed signal and then subtracts that from the received signal, leaving only the signal generated by the far end.



Quat	+3	+1	-3	-3	+1	+3	-3	-1	-1	+1	-1	-3	+3	+3	-1	+1	+1	+3
Bits	10	11	00	00	11	10	00	01	01	11	01	00	10	10	01	11	11	10

In each Quat the first bit is the sign bit and the second bit is the magnatitude bit

U interface 2B1Q coding
Figure 2

Transmission coding over the U interface is called two binary one quantary or ‘’**2B1Q**’’. Four voltage levels are used with each level representing 2 bits of the data being transmitted (fig. 2). Each of these two bit symbols is called a ‘’quat’’. The line rate for 2B1Q is 160K Bits/sec.

Problems, such as DC build-up and loss of timing, could arise if long strings of certain repetitive bit patterns were transmitted. To prevent this from happening scramblers are inserted at the telco CO as well as in the NT. In this way the DC build-up and clock recovery problems can be averted. These scramblers employ different polynomials, one specified for the CO line terminating unit (LT) transmitter and one for the NT transmitter. The use of different polynomials prevents like units from being placed back to back. For TAs that offer LDM mode one of the units must be able to employ the LT polynomial in its transmitter and the NT polynomial in its receiver.

The 2B+D data is sent over the circuit in a structure called a ‘’2B1Q Transmission Frame’’’. Like any TDM frame each 2B1Q transmission frame is a fixed number of bits and is repetitive. Each frame is 240 bits long and is formatted with three primary fields as shown in figure 3. Eight 2B1Q transmission frames comprise one 2B1Q superframe.

Sync Word	12 x 2B+D data	M Bits
Bits 1 - 18	Bits 19 - 234	Bits 235 - 240

Sync Word	18 Bits	9 2B1Q Quats	12000 Bits/sec
2B+D field	216 Bits	108 2B1Q Quats	144000 Bits/sec
M Bit field	6 Bits	3 2B1Q Quats	4000 Bits/sec
Total	240 Bits	120 2B1Q Quats	160000 Bits/sec

Frames in the NT-to-network direction are offset from
Frames in the network-to-NT direction by 60±2 Quats

2B1Q Transmission Frame
Figure 3

SYNC WORD The first 18 ‘’bits’’ of the 2B1Q frame is the Sync word. The Sync word is used to identify the start of a frame and the superframe. It should be noted that this is the only part of the 2B1Q frame that is not scrambled before transmission. (See fig. 4)

2B+D This is the user data and signaling bits. In each frame there are 96 bits of B1, 96 bits of B2, and 24 bits D of channel information. The format is 8 bits of B1, 8 bits B2, and 2 bits of D. This sub-format repeats 12 times per frame between the sync and M bits for a total of 216 bits/frame. (See fig. 4)

M BITS M bits are used for loop commands from telco, a frame CRC character, and status bits indicating both LT and NT status. This provides a method of monitor and control from the CO that is non-interuptive to the user. (See fig. 4)

Located in the last 6 bit positions of every 2B1Q transmission frame are the M bits. M bits are shown in figure 4. Some M bits are single status indicators such as the activation bit that is either a 1 (on) or a 0 (off). Other M bits must be read in groups over a series of frames. For example the CRC character encoded in Mbit positions 5 and 6 is 12 bits long and is calculated using the 2B+D bits and the M4 bit of every frame. It takes an entire superframe to calculate the CRC character.

Another important M bit function is the Embedded operating Channel or EOC. This field is used to send and receive status information and commands between the telco LT and the NT. Loops include a B1 loop, B2 loop, and 2B+D loop (fig. 5). A loop can be placed on one B channel while user data runs uninterrupted on the other B channel. 2B+D loops are, of course, interruptive. Three of the EOC bits are the address, one is the data (0) or message (1) indicator, and eight are the data or message itself. These 12 bits repeat twice every superframe for a total of 24 bits per superframe. M bits are considered an NT function and normally are not passed on the TE. In the case of a U interface TE the distinction becomes less clear as the unit contains an internal NT that supports all of the required functions.

Monitoring for errors is done through the CRC calculated at the superframe level. If the calculation results in identification of a superframe received with errors a notification bit called the FEBE is set and sent to the far end device in the next transmitted superframe. Once an LT or NT1 receives a FEBE bit it can use this information to set an alarm notifying the user. Most NT1 and LT devices count FEBE bits received and display them. Additionally most manufacturers present counts of the NEBE as well. The NEBE is not a defined bit but is instead a count of superframes received with invalid CRCs. The result is the ability to display both received errors (NEBE) as wells as errors received by the far end unit (FEBE). Typically NT1 manufacturers provide displays of this information for monitoring purposes.

Bit position	1-18	19-234	235	236	237	238	239	240
Quat position	**1-9**	**10-117**	**118s**	**118m**	**119s**	**119m**	**120s**	**120m**
Bit Name	**Sync Word**	**2B+D**	**M1**	**M2**	**M3**	**M4**	**M5**	**M6**
Frame 1	ISW	2B+D	eoa 1	eoa 2	eoa 3	act	1	1
Frame 2	SW	2B+D	eodm	eoi 1	eoi 2	ps 1	1	febe
Frame 3	SW	2B+D	eoi 3	eoi 4	eoi 5	ps 2	crc 1	crc 2
Frame 4	SW	2B+D	eoi 6	eoi 7	eoi 8	ntm	crc 3	crc 4
Frame 5	SW	2B+D	eoa 1	eoa 2	eoa 3	cso	crc 5	crc 6
Frame 6	SW	2B+D	eodm	eoi 1	eoi 2	1	crc 7	crc 8
Frame 7	SW	2B+D	eoi 3	eoi 4	eoi 5	sai	crc 9	crc 10
Frame 8	SW	2B+D	eoi 6	eoi 7	eoi 8	1	crc 11	crc 12

Eight Frames = One Superframe (s=Sign m=Magnitude)

2B1Q Transmission Frame Bit Position (NT to Network direction)

Figure 4a

Bit position	1-18	19-234	235	236	237	238	239	240
Quat position	**1-9**	**10-117**	**118s**	**118m**	**119s**	**119m**	**120s**	**120m**
Bit Name	**Sync Word**	**2B+D**	**M1**	**M2**	**M3**	**M4**	**M5**	**M6**
Frame 1	ISW	2B+D	eoa 1	eoa 2	eoa 3	act	1	1
Frame 2	SW	2B+D	eodm	eoi 1	eoi 2	dea	1	febe
Frame 3	SW	2B+D	eoi 3	eoi 4	eoi 5	1	crc 1	crc 2
Frame 4	SW	2B+D	eoi 6	eoi 7	eoi 8	1	crc 3	crc 4
Frame 5	SW	2B+D	eoa 1	eoa 2	eoa 3	1	crc 5	crc 6
Frame 6	SW	2B+D	eodm	eoi 1	eoi 2	1	crc 7	crc 8
Frame 7	SW	2B+D	eoi 3	eoi 4	eoi 5	uoa	crc 9	crc 10
Frame 8	SW	2B+D	eoi 6	eoi 7	eoi 8	aib	crc 11	crc 12

Eight frames = One Superframe (s = sign m = magnitude)

2B1Q transmission Frame Bit Position (Network to NT Direction)

Figure 4c

Inverted sync word	Quat pattern=+3+3-3-3-3+3-3+3+3	Delineates the first frame of the superframe
Sync Word	Quat pattern=+3+3-3-3-3+3-3+3+3	Marks the start of frames 2 through 8 of a superframe

The ISW and SW are the only non-scrambled information sent over the 2B1Q interface

2B1Q Sync Word Pattern

Figure 4c

	Channel B1				Channel B2				D ch.
Bit Pairs	b1 b2	b3 b4	b5 b6	b7 b8	b1 b2	b3 b4	b5 b6	b7 b8	d1 d2
Quat #	10	11	12	13	14	15	16	17	18

The above 2B+D pattern repeats 12 times per basic rate (2B1Q) frame and 96 times per superframe

2B+D Field of 2B1Q Frame (bits 19-234)

Figure 4d

(Cont. next page)

act = start-up bit	1 = Start-up, ready for Layer 2
aib = alarm indicator	0 = Interruption of circuit between NT and CO
crcX = cyclic redundancy check (use 2B+D bits and M4 bits)	crc1 = msb, crc12 = lsb
cso = cold start only	1 = Cold start only NT transceiver (most common type)
dea = turn-off bit	0 = Indicates network turning off circuit to NT
eoXX = embedded operations channel	bit pattern dependent upon message
eoaX = address	eoa1 = msb, eoa12 = lsb
eodm = information type	0 = Message, 1 = Data
eoiX = information	data or message information field
febe = far end block error	If RX has crc error then TX = 0
ntm = NT test mode	0 = NT in user test (i.e. Loop), 1 = NT normal
ps1, ps2 = power supply status	0 = PS problem, Unit must be able to send for 3 superframes after power failure
sai = S interface activation indicator	1 = Indicates S/T activity
uoa = U only activation bit	1 = Activate S/T interface (if present)

Note 1 All unused M bit positions should be set to 1 and are reserved for future use

Note 2 In normal operation the status bits should = 1 when there are no problems. One exception is ps2 because most NTs do not have a backup power supply.

M-bit Field of 2B1Q Frame (bits 235-240)
Figure 4e

Since only one NT can be connected to a Basic Rate circuit it may seem unnecessary to have an address field within the M bits, however, there are often other devices between the CO and the NT. Much as channel banks or repeaters exist in standard digital circuits similar devices exist in Basic Rate ISDN networks. For testing purposes NT1's always respond to address 7 (broadcast) and address 0 (NT). Figure 5 list the basic EOC commands.

Messages	Message Code	Message Direction
2B+D Loop	0101 0000	LT to NT
B1 Loop	0101 0001	LT to NT
B2 Loop	0101 0010	LT to NT
Request Bad CRC	0101 0011	LT to NT
Notify Bad CRC	0101 0100	LT to NT
Return to Normal	1111 1111	LT to NT
Hold State	0000 0000	Both
Unable to Comply ACK	1010 1010	NT to LT

EOC Message Coding and Direction
Figure 5

When a device is first attached to the ISDN U interface a pattern called *Wake up* is begun. During wake up the NT and LT exchange known quat patterns in order to train the equalizers and echo cancelers. The wake up pattern is a combination of the +3 and -3 quat patterns producing a 10 kHz signal. That is followed by a series of signals exchanged between the NT and the LT that signal startup and activation (see act bit) where the frame and superframe are recognized and in sync. At this point D channel information may be exchanged. Activation of a 2B1Q line can take up to 15 seconds. Activation of Layer 1 signals the NT and LT that Layer 2 can now be made active.

The Layer 1 U interface specification for ISDN Basic Rate is defined by the American National Standards Institute (ANSI). The ANSI T1.601 1992 document gives all the specifications used by the NT portion of all North American U interface Terminal Adapters, NT and LT devices. There is also an ITU version of the specification called G.961 and an ETSI standard called ETR 080. All versions of this standard are very similar.

The U interface has become the standard internationally for the loop between the telephone company central office and the user premises. In North America the user is responsible for purchasing and attaching the NT1 device to terminate the U interface. Often the NT1 is built into equipment such as routers and Terminal adapter.

Most of the rest of the world provides the NT1 as part of the BRI circuit. Many of these devices contain not only the U to S/T conversion but may include TAE functions. Additional NT1 functions may be ISDN telephone capabilities that allow the user to attach a standard 2 wire analog telephone to the NT1 without any other equipment required. Routing capabilities are also available in NT1 devices.

With the exception of North America most countries power the NT1 over the U interface. To accomplish this a DC voltage is applied to the same wire pair carrying the 2B1Q signal. This becomes important because it insures the NT1 is powered even when power is lost at the customer premises. Some applications allow the NT1 to go into a standby state, using less power, when no connections are active.

In North America the DC current applied to the U interface is only for the purpose of sealing current functions. Sealing current is applied to most circuits to prevent buildup of corrosion on cable splices and other connections. This current is not intended to power the NT1 and as a result in North America the power for the NT1 is provided locally by the end user.

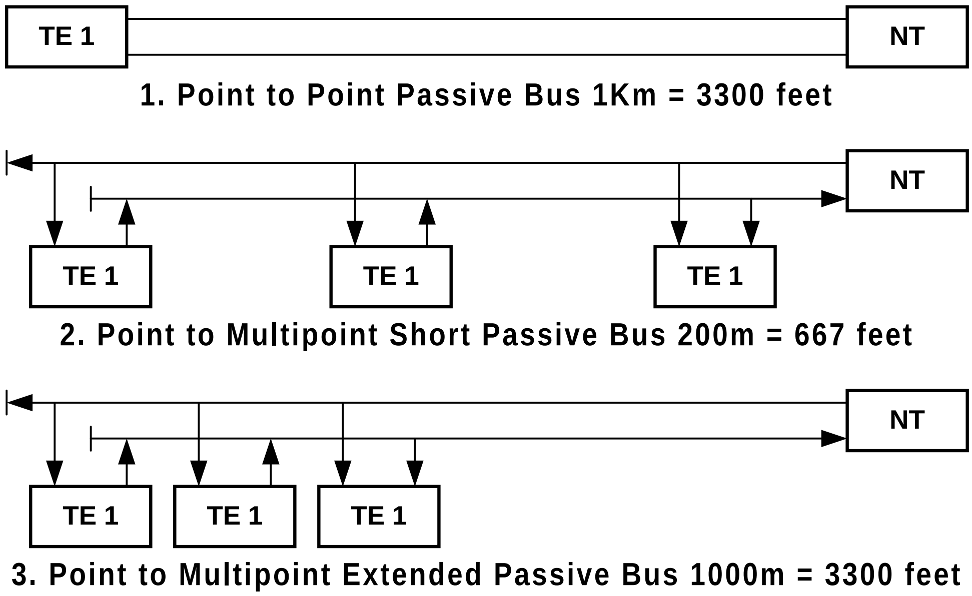
A growing trend has been seen to use the two-wire 2B1Q method for leased line service. In this mode the Telephone Company supplies a two wire U interface circuit at the customers central location and another at the customers remote site. This is now referred to a IDSL and comes in two versions. This provides the user with an increased speed over 64Kbps dedicated circuits without the need to move to a much more expensive T1 or E1 circuit.

Two different implementations are offered for IDSL service. One supports 144Kbps by including both B channels and the D channel. Because this is a dedicated point to point connection the D channel has no signaling functions. The second method of is to use only the B channels for 128Kbps service. Most TAs designed specifically for use with routers and other data only applications support this method of operation.

BRI Layer 1 "S/T" Interface

The S/T interface allows for two different types of bus arrangements, point to point and multipoint as shown in figure 6. Multipoint in ISDN terms does not mean the same thing as in the leased line modem or DDS world. Up to 8 S/T interface TEs or TAs can be placed in parallel across the S/T passive bus. When multiple TEs are on the bus only one unit can use a B channel at a time. As an example one TE can be using B1 while another TE can be using B2. Until one of these B channels is released no other TEs can have access to transmit or receive over a B channel.

This passive bus is a 4-wire type using one pair for transmit and one pair for receive. S/T interfaces operate at 192 Kbps with a line coding called ''PSEUDO-TERNARY SIGNALING''. In this coding a ONE bit is represented by 0 volts or no signal while a ZERO bit is a pulse of ±750 mV with the pulses alternating in polarity. NTs have the task of reformatting the data between the U and S/T interfaces.



Termination resistors can be internal (1) to the TE or external (2 & 3) as shown above.

1) Limited to one physical unit terminating the S/T circuit from the NT.
2) TEs can be placed at any point along the short passive bus.
3) All TEs must be within 50 meters of each other due to signal propagation time.

Point to Point and Point to Multipoint Passive Busses
Figure 6

Each pseudo-ternary signaling frame is 48 bits long and contains 36 bits of 2B+D data and 12 bits of overhead. The B channel and D channel bit interleaving on the S/T is 8 B1 bits, 1 D bit, 8 B2 bits, 1 D bit repeated twice per frame. Overhead bits are interspersed between the groups of B bits and D bits. The TE and NT use these overhead bits that include framing, activation, and DC balancing bits. Balancing bits are used to prevent DC buildup by making the number of + and - pulses equal. Figure 11 details the bit positions within the pseudo-ternary frame.

NT to TE distance in a point to point configuration is specified at 3300 feet (1 kilometer). In multipoint configurations limits are set at 667 feet (200 meters) when the TEs are randomly spaced along the bus. Multipoint Distances of 3300 feet can be reached if all TEs (or TAs) are clustered at the far end of the bus. The reason for these limits is that all TEs on the bus receive their clocking from the NT on the receive pair and must maintain transmit bit phase to function properly. All TEs will view each bit time in the frame as a bit time slot and will only transmit a D bit in the D bit time slot, a B1 bit in a B1 bit time slot etc.. Given too long a distance between TEs their transmit time slots might become out of phase with each other. The result would be that the NT would receive bits in the wrong time slot.

A major concern in this passive bus arrangement is the problem of contention. A D bit echo from the NT is monitored by the TEs to overcome this problem. Whenever a TE wants to send data on a B channel it must request use of the B channel by sending a message over the D channel. Before a unit can transmit on the D channel it must monitor the D channel echo bit to see if any other TE is transmitting. The TE will count receive echo bits, and if it sees all ones (0 volts) for 8 bit times it may begin transmitting. If two units begin transmitting a 0 bit simultaneously they will both see a 0 bit echoed, if they both transmit a 1 bit they will see a 1 bit echoed, however, when one unit sends a 1 and the other sends a 0 the echo will be a 0. When a unit sends a 1 (0 volts) and sees a 0 (.75 volt) echoed back it stops transmitting while the unit that sees a proper echo bit continues to transmit. In this way the passive bus operates as a logical AND device. Because the TEs and TAs are on the bus in parallel if multiple units transmit a 0 the voltage representing the 0 is not additive.

In the event of high demand for D channel accesses a priority scheme is needed to prevent a unit from being locked out for an extended period of time. After successfully transmitting on the D channel a unit must wait more than 8 echoed 1 bits before transmitting again. When a long string of 1 bits is detected the priority counters are reset.

Due to the frame offset between the TE receive and transmit directions the echo bit is seen before the next D bit is transmitted allowing for immediate recognition of contention. Contention for the D channel is not a problem at the U interface because only one physical device is connected.

The NT derives its timing from the 2 wire U interface and must be slaved to that source. The TE must be slaved to the NT. In this way the telco Line Terminator Unit in the CO provides timing and synchronization to all devices on the Basic Rate S/T interface Circuit.

S/T interface specification contains two optional channels that allow diagnostic and status information to be passed between a TE and the NT. The two channels are called the Q and S channels with the S channel being divided into two sub-channels SC1 and SC2. The following is a list of these channels and their functions. While provided for in the specification it is not mandatory that TE and NT equipment support these functions, and many do not.

Message Name	Description of the message
Idle	Unit is in normal operational mode
Power Loss	TE has lost power, sent for 1 to 3 frames
S/T Request	Request the NT to do a self test (see SC1 channel S/T Report responses)
Loop B1	NT Loop channel B1 back to the TE (see SC1 channel LB1I loopback indicator)
Loop B2	NT Loop channel B2 back to the TE (see SC1 channel LB2I loopback indicator)
Loop B1 and B2	NT Loop both channels B1 and B2 back to the TE (see SC1 channel LB1/2I loopback indicator)

Q Channel TE to NT direction
Figure 7

In the S/T frame format 20 frames make up one multiframe with the M bit used to delineate the first multiframe. The Q and S channel commands and messages are encoded into the S/T frame with each command or message comprised of four bits. Q bits can be found in the FA bit position of frames l, 6, 11, and 16 of the S/T frame from the TE to the NT. SC1 bits are in the FA bit position of frames 1, 2, 3, and 4 in the NT to TE direction. SC2 bits are in the FA bit position of frames 5, 6, 7, and 8 in the NT to TE direction. Bit positions 9 through 20 are not used at present but are reserved as follows FA bits 9-12 = SC3, bits 13-16 = SC4, bits 17-20 = SC5.

Message Name	Description of the message
Idle	Unit is in normal operational mode
Power Loss	TE has lost power, sent for 1 to 3 frames
S/T Request	Request the NT to do a self test (see SC1 channel S/T Report responses)
Loop B1	NT Loop channel B1 back to the TE (see SC1 channel LB1I loopback indicator)
Loop B2	NT Loop channel B2 back to the TE (see SC1 channel LB2I loopback indicator)
Loop B1 and B2	NT Loop both channels B1 and B2 back to the TE (see SC1 channel LB1/2I loopback indicator)

Q Channel TE to NT direction

Figure 8

The Q channel is used in conjunction with the SC1 channel. When a command such as a loopback is issued from the TE to the NT over the Q channel the response from the NT will come back over the SC1 channel. Figures 7 and 8 list the Q channel commands. Figure 9 show the SC1 sub-channel messages.

The SC2 channel is primarily used to provide a method of passing U interface status, alarm and test information found in the M bits and EOC channel of the U interface 2B1Q frame from the NT to the TE. There is no collision detection mechanism or addressing on the Q and S channels. Figure 10 shows the SC2 channel messages.

Message Name	Description of the message
Idle	Unit is in normal operational mode
Power Loss	TE has lost power, sent for 1 to 3 frames
STP	NT has passed the self-test requested by the Q channel S/T request command
STF	NT has failed the self-test requested by the Q channel S/T request command
STI	NT is in the self-test mode this may be as result of the Q channel S/T request command
FEC	Far end code violation, last frame received from the TE(s) had a violation of the S/T line code format
DTSE-IN	Detected access transmission system error. This code is passed from the NT to the TE when the NT receivesan error in the frame from the LT. See NEBE
DTSE-OUT	Detected access transmission system error. This code is passed from the NT to the TE when the LT receivesan error in the frame from the NT. See FEBE
DTSE-IN/OUT	Tis is a combination of the above conditions. If the reason for the transmit and receive errors is an EOC corruption of CRC command the DTSE will be suppressed.
LB1I, LB2I, LB1/2I	These are the indicators that the NT is doing the loop requested by the TE over the Q channel.
LRS	Loss of receive signal indication is sent to the TE when the NT can not identify the signal being received on the U interface.
DOI	Disruptive NT operation indication is sent to the TE when the D channel information flow is not normal. An example of this is if the NT is in a 2B+D loop.

SC1 Subchannel NT to TE direction

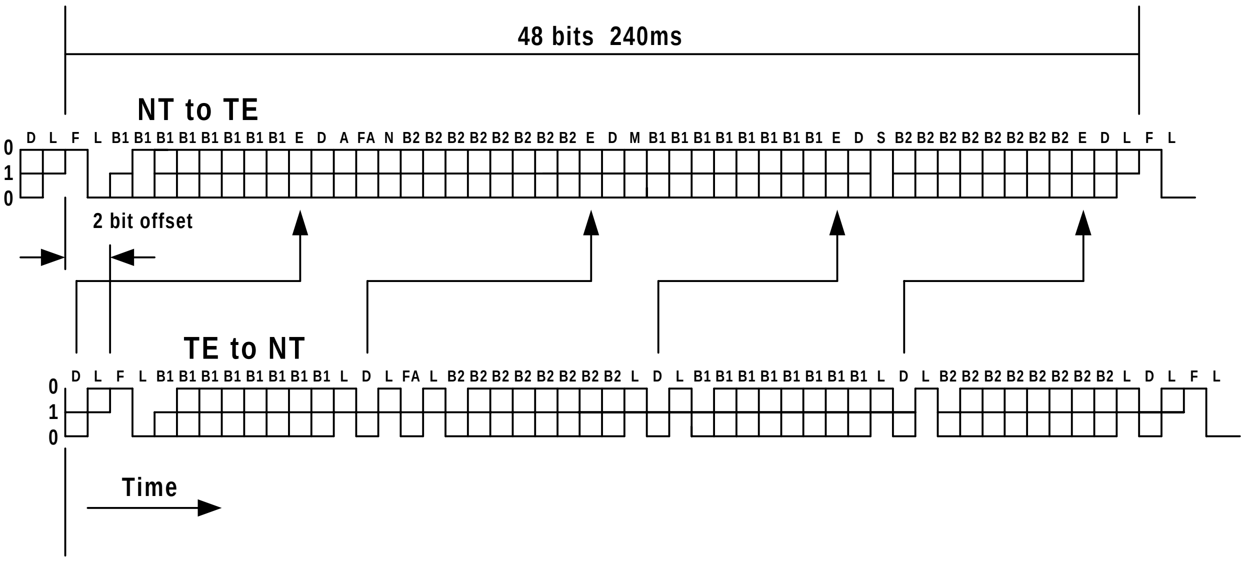
Figure 9

The Q and S channels are only defined by ANSI as optional for use in North America. Most Basic Rate ISDN equipment does not support the Q and S channels. Q and S channel support is uncommon and is included for reference purposes.

Message Name	Description of the message
Rdea	U interface DEA bit = 0 (see U interface M bits)
Ract	U interface ACT bit = 0 (see U interface M bits)
NAI	Network has notified NT of a network failure (see U interface aib bit)
PPb	Primary power bad (see U interface M bits)
PSb	Secondary power bad (see U interface M bits)
LUTI	NT1 in 2B+D loop or quit mode (see U interface EOC commands)
LB1N	B1 looped back to network (see U interface EOC commands)
LB2N	B2 looped back to network (see U interface EOC commands)
CcrcR	Corrupt CRC requested (see U interface EOC commands)
CcrcN	Corrupt CRC notified (see U interface EOC commands)
Rset	NT has gone into reset state (i.e. U interface resync)
Rueoc	NT received unknown EOC command or message
IBNAM	Implemented but No Active Message is the normal message transmitted when the U interface is up and there are no alarms or test
Extend	Introduce Extended set of messages (I have no clue as to what the extended messages are or how they are used)
SNI	Subchannel not implemented is sent when the SC2 channel is not implemented in the NT1

SC2 Subchannel NT to TE direction

Figure 10



F Framing bit
L DC balancing bit
D D channel bit
E D channel echo bit (NT to TE)
FA Auxiliary framing bit
M Multiframing bit
N Set opposite of FA (NT to TE
B1 User data channel B1
B2 User data channel B2
A Activation bit
S Reserved for future use

ST Passive Bus Frame Format 192 Kbps

Figure 11

	S/T	U
Standards	ANSI T 1.605 1991 ITU-T I.430 ETSI ETS 300 012	ANSI T 1.601 1992 ITU-T G.961 Annex A ETSI ETR 080
Max Cable Run	1 KM / 3300 ft.	5.5 KM/18K ft
Signal Loss	6 dB @ 96kHz	42 dB @ 40 kHz
N America Connector	RJ45	RJ45
N America NT RX Pair	Pins 3&6	Pins 4&5
N America NT TX Pair	Pins 4&5	TX and RX on 1 pair
International Connector	Country Specific	Country Specific
Location	Indoors only	Indoor or Outside
Wire Pairs	2 (1 Tx 1 Rx)	1 Echoed Canceled
DC Balancing	Balancing Bits	Scrambled
Total Line Rate	192 KBPS	160 KBPS
Line Frequency	96 kHz	40 kHz
N America DC current	Optional TE power feed on bus	Sealing current use only
International DC current	Optional TE power feed on bus	NT1 powered from CO

Comparison of S/T interface and U interface

Figure 12

Figure 12 provides a comparison of the differences between the S/T and the 2B1Q U interface.

Primary Rate ISDN Physical Interface

PRI is available in two types 23B+D and 30B+D. Available offering in any particular country depend upon the telephone standards used. 23B+D is based on the standard 1.544 Mbps T1 carrier technology deployed in North America, Japan and Korea. 30B+D is found in countries that use the standard 2.048 Mbps E1 carrier technology found in most of the rest of the world.

The most obvious difference between the two methods of PRI has to do with the number of B channels supported on a single interface, 23B vs. 30B. This difference results from the difference between T1 and E1 carrier standards that predate ISDN. PRI was designed to use these existing technologies for layer one transport with no substantive changes. There are many reference documents available on these topics so I have not included detailed information, only a basic overview.

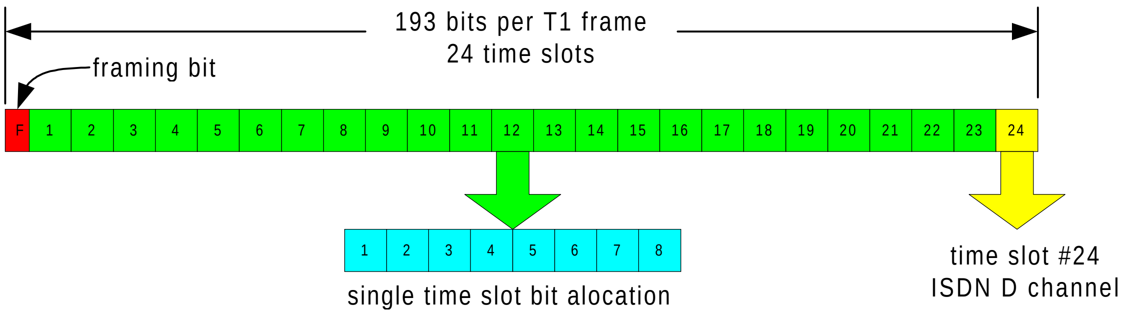
23B+D PRI

As previously mentioned the 23B+D PRI interface uses T1 for the layer one physical transport. While T1 is available in D4-AMI format when PRI is involved the framing format must be Binary 8 Zero Substitution (B8ZS) with Extended Super Frame (ESF).

With T1 only 1 bits cause a pulse on the circuit. As a result a long string of zeros could cause errors due to a loss of timing. B8ZS is used to insure that long strings of zeros do not result in loss of timing by substituting a special pattern of one and zero bits to represent a group of zeros.

T1 provides for 192 data bits (24 time slots x 8 bits) preceded by 1 framing bit resulting in a 193-bit frame. 8 consecutive frames equal one superframe and 3 superframe equal one extended superframe.

Of the 24 time slots one must be reserved for D channel signaling. On a single PRI this will always be time slot 24. The other time slots (1-23) are the B channels where the user data will be placed. This works very much like BRI except there are 23 available B channels instead of only 2. Figure 13 shows the basic T1 frame that is used in 23B+D PRI.



PRI 23B+D frame format

Figure 13

The Tl frame repeats 8000 times per second. At this rate each 8-bit time slot (B channel) is seen 8000 times per second. The result is the 64 Kbps B channel speed. The D channel also operates at 64 Kbps. The functions of the B and D channels are the same as with BRI, only the number of B channels and the speed of the D channel are different.

The first bit of each frame is called the F bit. The function of the F bit is dependent upon the position of the T1 frame within the ESF. Starting with frame number four an alignment pattern is formed using every fourth frame. This repeating pattern is 001011.

Starting with the first frame every other frame uses the F bit for the facilities data link. This allows a monitor, test and maintenance channel for use by the carrier. Information on alarms and status in encoded using these bits. For testing purposes loop commands can also be encoded into these bits as well.

Starting with the second frame and continuing with every fourth frame the F bit is used to encode a CRC-6 value for error checking. The CRC is functional at the ESF level and can be used to insure the integrity of the physical layer. Figure 14 provides a list of F bit assignment within the T1 ESF frame.

T1 Frame Number	Extended Super Frame F Bits assignments		
	Framing bits	data Link bits	CRC bits
1	-	m	-
2	-	-	e_1
3	-	m	-
4	0	-	-
5	-	m	-
6	-	-	e_2
7	-	m	-
8	0	-	-
9	-	m	-
10	-	-	e_3
11	-	m	-
12	1	-	-
13	-	m	-
14	-	-	e_4
15	-	m	-
16	0	-	-
17	-	m	-
18	-	-	e_5
19	-	m	-
20	1	-	-
21	-	m	-
22	-	-	e_6
23	-	m	-
24	1	-	-

F bit use in ESF T1

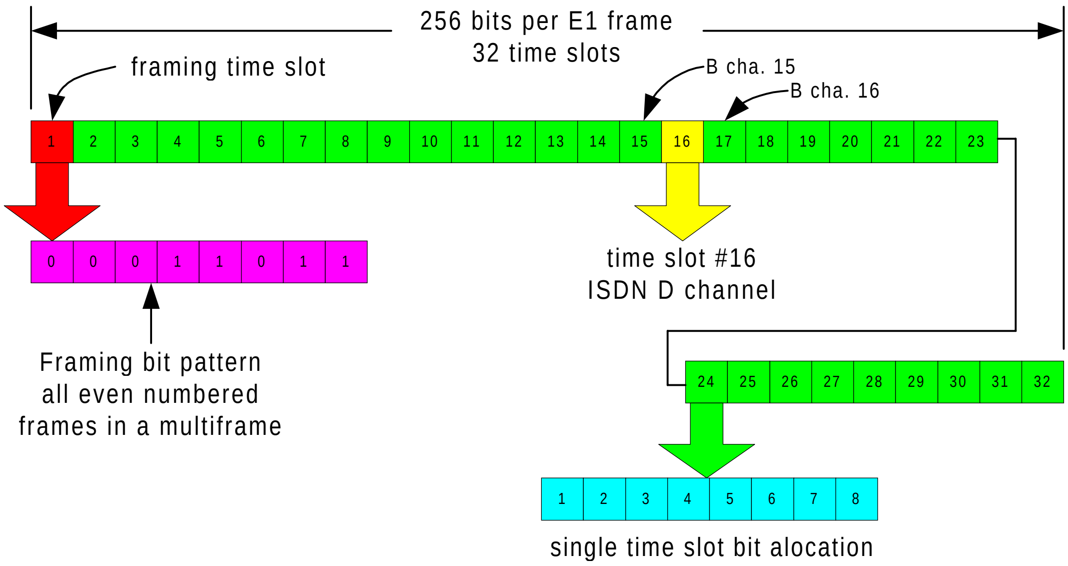
Figure 14

30B+D PRI

30B+D PRI is based on standard E1 carrier technology common around the world. E1 uses a format called HDB3 to insure excess strings of zeros do not result in loss of clocking. This is analogous to the function that B8ZS serves in T1 based PRI. Information on HDB3 can be found in a number of sources and will not be detailed here.

Each E1 frame contains 32 time slots of 8 bits each for a total frame length of 256 bits. Of these 32 time slots 30 are available for user data in the form of B channels. Time slot 1 is reserved for frame alignment and optional CRC, monitor and alarm bits the same as FAS in non-PRI applications. Time slot 16 is reserved for the D channel. In non-PRI applications time slot 16 is used for channel associated signaling or CAS.

The frame structure for 30B+D PRI is shown in figure 15. Frames repeat at the rate of 8000 frames per second resulting in 64 Kbps per time slot and 2048 Kbps for a single frame. Because the D channel is located in time slot 16 the B channel numbering is as follows. B channel 1 is in time slot 1 this continues on a one for one match up to time slot 16. B channel 16 is in time slot 17 and B channel 30 is in time slot 31.



PRI 30B+D frame format
Figure 15

Multiple E1 frames are arranged to form an 8-frame sub-multiframe (SMF) and a 16-frame multiframe (MF). Starting with the first frame and continuing with every other frame time slot one contains the bit pattern C0011011 to identify the start of a frame. The C represents a single bit of the 4 bit CRC-4. The CRC is functional at the SMF level and requires 8 frames to determine an error condition.

Sub-multiframe	Frame Number	Bits 1 - 8 of each frame in the Multiframe							
		1	2	3	4	5	6	7	8
1	0	C1	0	0	1	1	0	1	1
	1	0	1	A	S4	S5	S6	S7	S8
	2	C2	0	0	1	1	0	1	1
	3	0	1	A	S4	S5	S6	S7	S8
	4	C3	0	0	1	1	0	1	1
	5	1	1	A	S4	S5	S6	S7	S8
	6	C4	0	0	1	1	0	1	1
	7	0	1	A	S4	S5	S6	S7	S8
2	8	C1	0	0	1	1	0	1	1
	9	1	1	A	S4	S5	S6	S7	S8
	10	C2	0	0	1	1	0	1	1
	11	1	1	A	S4	S5	S6	S7	S8
	12	C3	0	0	1	1	0	1	1
	13	E	1	A	S4	S5	S6	S7	S8
	14	C4	0	0	1	1	0	1	1
	15	E	1	A	S4	S5	S6	S7	S8

PRI 30B+D Multiframe format
Figure 16

A multiframe is defined by a bit pattern found in the first bit position of each frame in the multiframe. The first bit position alternates between the MF alignment bits and CRC related bits. The pattern repeats C0C0C1C0C1C1EE. The C and E bits are CRC related. The remaining 0-0-1-0-1-1 provides for MF alignment. Figure 16 shows the structure of the first timeslot of an E1 PRI frame.

The C bits represent the CRC-4 bits and are used to identify bit errors at the SMF level. CRC support is optional in E1 and as a result users caution should be used when setting up and testing circuits.

At the MF level the E bits are used to identify CRC detected errors in SMF. When a CRC error is detected in a single received SMF an E bit, on a transmitted frame is set to 0. Two E bits are provided to identify problems with each SMF of the MF.

The remaining bit positions are S bits and A bits. The S bits are designed to provide a variety of optional capabilities such as monitor, maintenance and control functions. Because they are optional each country may define a unique use for these bits. The A bit is a remote alarm indicator bit used to identify the remote end of alarm conditions.

Layer 2

The Data Link Layer

Layer 2 called the data link layer. In ISDN it is called the **L**ink **A**ccess **P**rocedures **D C**hannel or **LAPD**. It is the purpose of the LAPD protocol to provide a framework for transferring data with the ability to ensure that the data is error free. The 5 fields of LAPD are shown below. Figure 17 shows a block diagram of the LAPD framework.

FLAG The first and last 8 bits of every LAPD frame is the flag. The flag character is always 01111110 (hex 7E).

ADDRESS Following the starting flag is the 16 bit (2 octet) field containing the Service Access Profile Identifier or SAPI, and the Terminal Endpoint Identifier or TEI.

CONTROL There are three types of frames defined within LAPD. It is the job of the control field to inform the receiving device (TE or LT) what type of information is being transmitted in this frame. There is also a Next Send and Next Receive (NS, NR) frame numbering system.

INFO The information field is where the layer three information will be placed. This field is of variable length.

FCS The last 16 bits before the ending flag of a LAPD frame is the Frame Check Sequence or FCS.

Because data in the information field could contain a flag pattern, a 0 is inserted after every five l's. Only the flag is not sent through the 0 inserter. The receiving end looks for the flag pattern, all data that is not a flag pattern is sent through a 0 deleter to restore the original bit pattern. The deleter will remove every 0 that is preceded by five 1's.

BRI is designed with multiple TEs at the customer site all sharing the same basic rate circuit. It becomes important that the telephone switching equipment be able to address each individual TE. Referring back to figure 6 it is can be seen that each device on a basic rate circuit must have its own address.

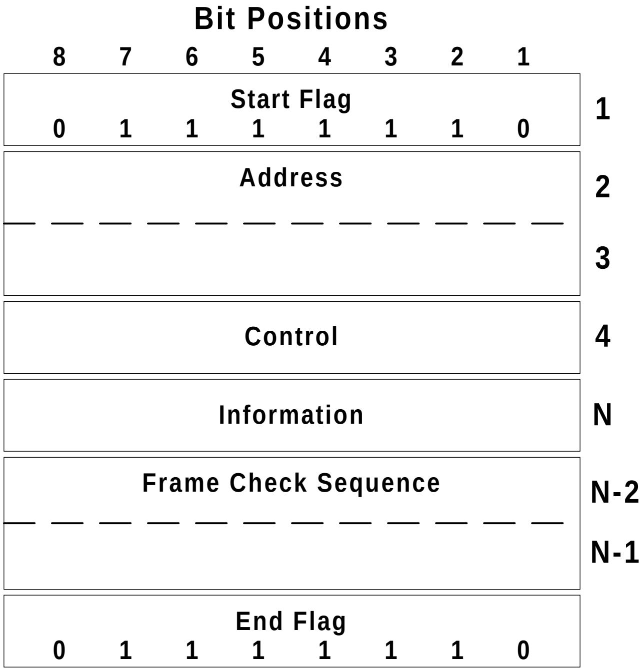
Addressing in LAPD is done in the two octets immediately after the starting flag. The address is split into two fields that, when taken together, direct the frame to the proper logical connection point.

The Service Access Profile Identifier or SAPI is the first part of the address. The SAPI is used to direct the information in the LAPD frame to the proper logical entity. A logical entity can be viewed as a software module that controls a set of functions.

An example of functions would be initiating a call, answering a call, and disconnecting a call. All messages dealing with call control of switched circuits would have the same SAPI, thereby directing the message to the proper control entity.

Figure 18 shows a list of the ITU SAPI values. This list may not be complete because of ongoing revisions by the ITU. All SAPI values used by terminal Adapter manufacturers are included in the list.

SAPI value 0 is used whenever a circuit switched voice or data call is requested or received. What this means is that when a TE wants to place a call, it will send all information needed for that call with a LAPD frame address of SAPI 0. Likewise when a call is incoming the message will use SAPI 0.



LAPD Frame Format
Figure 17

SAPI value 63 is used when the data in the information field of the LAPD frame is used for functions such as status and automatic TEI assignment. These two functions are Layer 2 management functions. X.25 equipment uses SAPI 63 for Layer 2 Management and SAPI 16 for the X.25 over the D channel.

SAPI VALUE	LOGICAL ENTITY
0	Call Control Procedures Layer 3
1	Packet Mode Q.931 Call Control
16	X.25 Packet Mode D Channel
63	Layer 2 Management

SAPI Values
Figure 18

The Terminal End Point Identifier is the number assigned to the TE at the customer premise by the local telephone company switch. There are provisions within ITU for both Fixed and Automatic TEI assignments. With the fixed method the telco CO switch is configured with one or more TEIs that must be configured in the TE. Fixed TEI addresses are most commonly used with X.25 applications.

Most Circuit switched ISDN applications use the automatic method. In the automatic method a TE requests a TEI assignment from the switch. The switch is free to assign any address between 64 and 126 that is presently unused on that circuit. ISDN terminal adapters will request a TEI when attached to the ISDN circuit. Figure 19 lists all valid TEI addresses and their uses.

PRI circuits differ from BRI circuits in that they don't use the automatic TEI procedures. One purpose of the TEI address is to identify individual physical units on a BRI multipoint circuit. Because PRI only operates on a point to point mode there is never more than one physical device. Because of this PRI devices use TEI 0, one of the fixed TEI addresses.

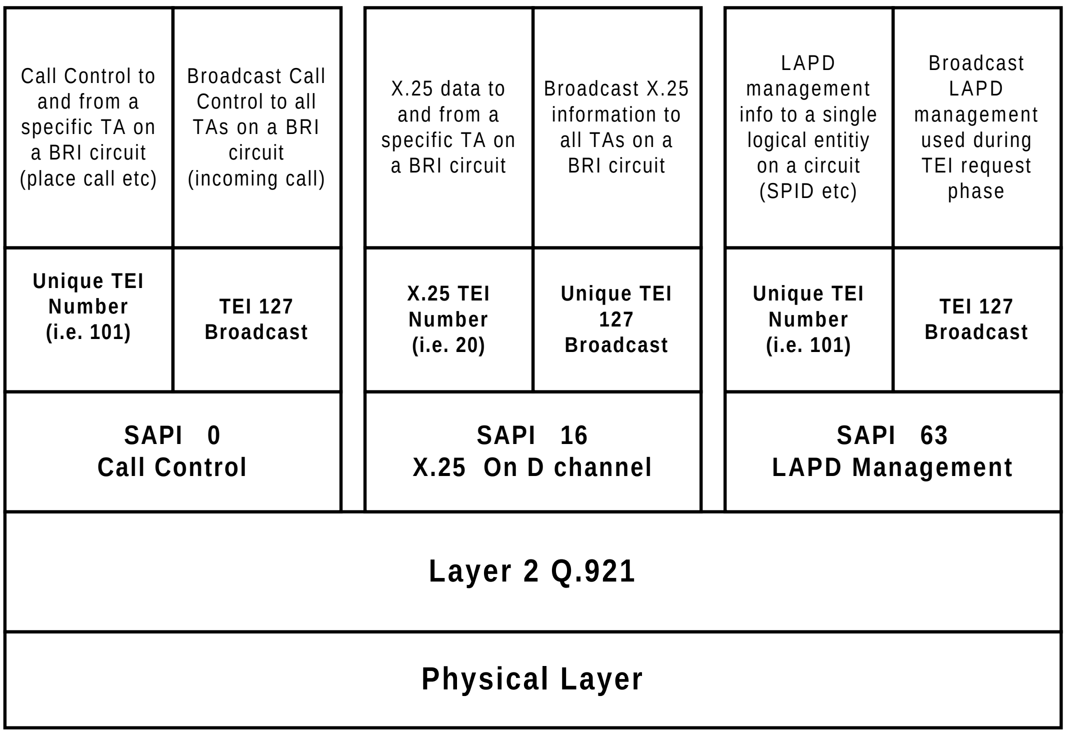
TEI VALUE	TEI TYPE
0-63	Non Automatic (fixed) TEI
64-126	Automatic TEI
127	Broadcast TEI

TEI Address Assignments
Figure 19

Together the SAPI and TEI are used to identify a logical connection point or software entity. This connection point is called the Data Link Connection Identifier or DLCI. Figure 20 shows the DLCI logical connections between the Telco CO and a TE.

Address assignments in LAPD have significance only between the TE and the LT. LAPD frames have no direct end to end functions. That is, the LAPD frame is not passed from the calling unit to the called unit directly. A call request is converted to Signaling System Seven (SS7) by the Telco switches for inter-switch communications and converted back to LAPD by the last switch for transmission on the ISDN line serving the called number. It is a requirement of true end to end ISDN service that all central office switches be SS7 compatible.

Typically in the North America the ISDN signaling information is placed in the ISDN Users Part (IUP) of an SS7 frame. In the rest of the world where ETSI standards are employed the ISDN signaling is placed in the Telephone Users Part (TPU). This is also used for standard analog telephone signaling information.



Logical DLCI Connection Points for SAPI 0, 16 and 63
Figure 20

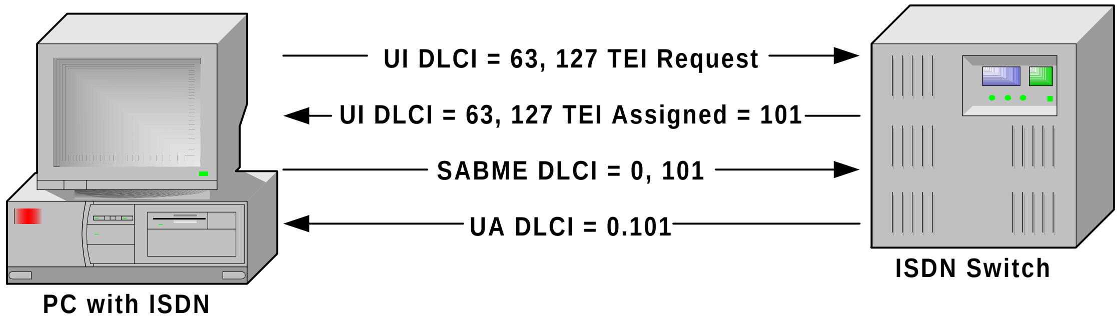
In Figure 20 it can be seen that incoming LAPD frames will first encounter layer 2. If the address has a SAPI of 0 and a TEI of 101 the frame will be passed to the call control entity. Although the diagram above only shows one logical call control entity, TEI 101, it is common for one Basic Rate Circuit to have multiple call control entities. If a terminal adapter contained X.25 packet mode capabilities there would be an X.25 packet switch control entity for that function with a SAPI of 16.

Also in Figure 20 no distinction is made between a TE and the switch. The porting of messages to the appropriate software module applies equally to the Central Office switch and the software in the TE. The difference is the jobs that the software at each is expected to perform. For example a TE will use SAPI 63 TEI 127 to request a TEI address while the switch will respond to the request over SAPI 63 TEI 127 with a TEI address in the info field of the response frame.

When the LAPD frame uses the SAPI 63 the frame is directed to the layer 2 management entity. Multiple entities with unique TEI addresses also exist in the Q.921 layer 2. Terminal Adapters in North America will generally request 2 TEI addresses, one for each phone number.

The use of a broadcast TEI (127) is to send a message to every TE on a circuit. If a Frame carries the DLCI of 63.127 then all layer 2 management entities on that basic rate circuit will receive and process the message. If a message has a DLCI of 0.127 all call control entities will receive and process the message.

The Automatic TEI addresses assigned by the Central Office switch or PBX LT must be maintained in a table within the switch. It is through the TEI addresses that the switch keeps a record of all units on a given basic rate circuit. Every time a device is disconnected and reconnected to the circuit a new address may be requested by the unit and assigned by the switch. To prevent running out of addresses the switch will poll units for their addresses periodically and may remove and assign a new address. If, after several attempts, no units respond to the TEI removal command the switch removes it from the poll table of currently active TEI addresses and places it in the available address pool.



LAPD TEI Assignment
Figure 21

Shown in figure 21 is the information exchange that takes place between the TE and the CO to assign a TEI address. First the unit requests an address with a UI frame. The switch response to this is another UI frame containing, in its information field, an address (101 in the example). Once a unit has an address it can request a logical connection for information transfer of layer 3 call control messages. This is done with the SABME frame. Once this has been acknowledged the unit is ready to exchange layer 3 (Q.931) information frames. Units that require multiple TEI addresses will perform the request for each address that they need and then send the SABME for each address assigned.

After the Flag and Address fields comes the Control field. Depending on the frame type the control field may be either 8 or 16 bits long. The Three types of LAPD control fields are shown in figure 22.

In figure 21 the TEI request and assignment process is shown taking place between a PC with an internal ISDN TE and the Telephone Company central office switch. If the PC were using a standalone TA attached via a COM port it would be the TA that would initiate the TEI request. The PC would have no knowledge or control of the process.

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
0	NS							P	NR						

Information Frame Format

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1	0	S	S	0	0	0	0	P/F	NR						

Supervisory Frame Format

1	2	3	4	5	6	7	8
1	1	M	M	P/F	M	M	M

Unnumbered Frame Format

S = Supervisory function bit
P/F = Poll / Final
M = modifier function bit

NS = Send Sequence Number
NR = Receive Sequence Number

LAPD Control Field Format
Figure 22

Information transfer or ‘’I’’ frames are used to transfer Q.931 (layer 3) or X.25 on the D channel messages. Only this frame uses the NS and NR fields. The NS field is the number of the frame being transmitted while the NR is the number of the next frame expected. Using this system allows the TE and the LT to monitor for missing frames.

Supervisory or "S" frames control the exchange of I frames. S frames are used to acknowledge I frames, transmit flow control information, and request retransmission of missing I frames based upon the NS-NR system. There are three types of S frames determined by the ‘’S’’ bits.

Unnumbered or ‘’U’’ frames come in seven types. U frames are used to establish and disconnect logical connections, negotiate data link parameters, and indicate errors that are not correctable via retransmission. No NS-NR sequence numbering is used in U frames. The ‘’M’’ bits determine the type of U frame.

Frame Type		C/R	Name	Description
I	I	C	Information	Used to transfer Q.931 (layer 3) data
S	RR	C/R	Receiver Ready	Used to ACK an I frame and to indicate ready to receive another frame
	RNR	C/R	Receiver Not Ready	Indicates a temporary inability to receive I frames
	REJ	C/R	Reject	Request I frame retransmission when an out of sequence I frame is received clears RNR condition
U	SABME	C	Set Asynchronous Balance Mode Extended	Request establishment of a Layer 3 connection between the switch and the customers TE equipment
	DISC	C	Disconnect	Terminates a logical link established by SABME
	UI	C	Unnumbered Information	Transfers link and layer 3 information that needs no ACK or sequence number
	UA	R	Unnumbered Acknowledgment	Acknowledges a SABME or a DISC
	DM	R	Disconnect Mode	Sent when a station has an error and cannot continue data transfer
	FRMR	R	Frame Reject	Sent when an error exists that cannot be cleared by retransmissions of a frame
	XID	C/R	Exchange ID	Intended for automatic data link layer parameter negotiation but not implemented to date

(c = command R = response)

LAPD Frame Types and functions

Figure 23a

Frame		Encoding	Frame		Encoding
I	I	0+7 bits NS - P+7 bits of NR	U	UI	1100P000
S	RR	10000000-X NR		UA	1100F110
	RNR	10100000-X NR		DM	1111F000
	REJ	10010000-X NR		FRMR	1110F001
U	SABME	1111P110		XID	1111X101
	DISC	1100P010			

(X=Poll/Final P=Poll F=Final) (NS=Next Send NR=Next Receive)

LAPD Control Field Bit Encoding

Figure 23b

Listed in figure 23a are all ITU defined layer 2 (Q.921) frame types. Figure 23b shows the bit encoding for the layer 2 frame types. All LAPD frames are defined as either Commands and or Responses (see figure 14) with a Poll/Final bit. Command frames use the poll bit while response frames use the final bit. Command frames set the poll bit to 1 to demand a response. Setting the final bit to 1 indicates a response to a command frame.

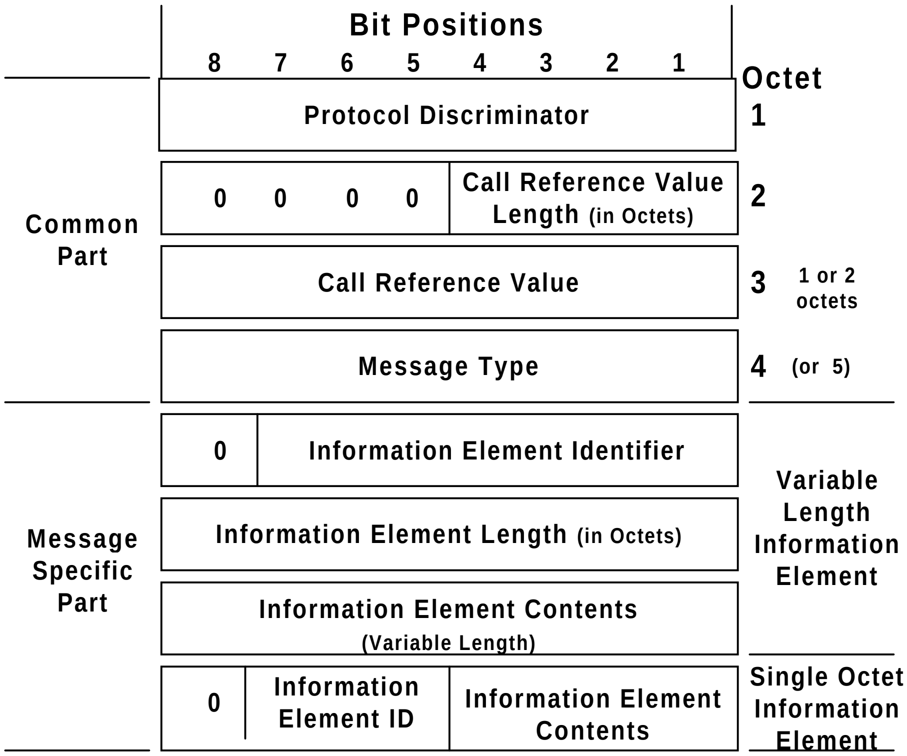
Layer 3

The Network Layer

Layer 3 or the Network Layer is defined by the ITU-T in the Q.931 recommendations. Within the specifications are protocols for establishing, maintaining, and clearing connections over Basic and Primary Rate ISDN using a bit oriented protocol. This information is transported over the D channel only.

In essence Q.931 is a list of messages that are exchanged between the users equipment, in this case a terminal adapter, and the local telephone company switch to which it is connected. Every Q.931 message has two basic parts. The first part or *Common Part* is at the head of every message. Four octets are taken by the Common Part as shown in figure 24. The first field is the protocol discriminator that indicates the message is a Q.931 call control message.

The second field is the Call Reference. This is a number that is assigned to a given call allowing multiple calls to be controlled by a single D channel. The call reference number only has significance between the TE and the local Telco switch and may be one or two octets long.



ITU Q.931 Frame Format
Figure 24

The third field of the common part is the Message Type. Message type refers to the actual function of the message, that is, does this message concern placing a call, disconnecting a call, etc. The sub-fields that follow in the second part of the Q.931 message further define the message type.

Q.931 Call Establishment Messages

Setup messages are sent to the switch by the TE when a call is placed. They are also sent from the switch to the TE when the TE is being called. Included with the SETUP message will be Information Elements to define the connection type being requested. In the case of a TE making a data call this will be a circuit mode, 64K, unrestricted data connection. The phone number being dialed will also be included in the Information Elements.

Call Proceeding is sent from the switch to the TE placing the call to indicate that all needed information has been received to set up the call. Information Elements associated with this message will tell the TE which B channel will be assigned to the call.

Alerting is sent from the called unit to the switch and from the switch to the calling unit to inform them that the called unit is ‘’ringing’’.

Connect is sent by the called unit to the switch when it answers the call and by the switch to the calling unit when the far end answers the call.

Connect Acknowledge is sent by both the TE and the switch to acknowledge the connect message. When this point is reached the units involved in the call are free to pass data over the B channel assigned by the switch to this call.

Q.931 Call Clearing Messages

Disconnect is the message sent by the TE or switch to terminate the call. Information Elements follow the disconnect message to Identify the type of disconnect. This may be normal call clearing or some error message.

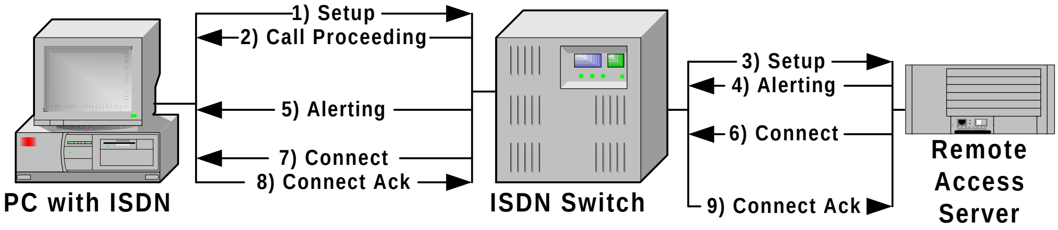
Release is a single octet message sent in response to a Disconnect. This message informs the receiving end that the B channel connection has been terminated.

Release Complete is the single octet message sent in response to the Release message. After this message is received the switch and TE are free to place another call.

Q.931 Miscellaneous Messages

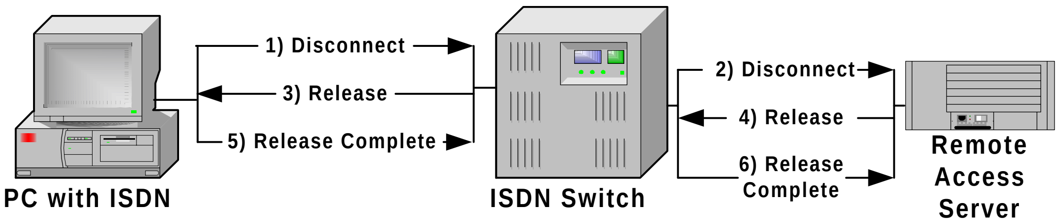
Status Enquiry is a request sent by the switch to determine the call status of the TE. This message is a single octet and contains no Information Elements.

Status is the response of the TE to a Status ENQUIRY. This is used to tell the switch the present status of the unit and if any calls are active. Status responses contain Information Elements.



Call Setup Message Exchange
Figure 25

Shown in figure 25 is the sequence of message exchange that takes place during a call setup. It is assumed that the call was completed without problems. Depending upon the equipment involved and the type of call there may be variations in the actual messages sent and received. After the call is active the D channel has no mandated requirement to remain directly involved in the connection.



Call Clearing Message Exchange
Figure 26

Figure 26 shows the sequence of messages exchanged during the disconnect phase. While the messages are shown having end to end relationships only the Disconnect message is propagated from source to destination. The release and release complete messages are not end to end and are exchanged between the local switch and user equipment once a disconnect message has been sent or received.

1) The dialing unit sends a **Setup** to the local switch. The information elements of this message will contain, among other things, the following.

   - Call Type (Voice, Data, X.25, etc.).
   - Circuit or Packet switched.
   - Data transfer rate 64K, 56K, (V. 110 Rate Adaption).
   - B-channel to use (B 1, B2, or switches choice).
   - Telephone number being dialed.

2) The switch responds with **Call Proceeding** indicating that the setup contained enough information to place the call and to indicate which B channel (l or 2) to use.

3) A **Setup** is sent to the called unit containing the call parameters requested by the calling unit.

4) The called unit sends **Alerting** to the switch to acknowledge the Setup.

5) **Alerting** is also sent to the calling unit to inform it that the remote is ''ringing''.

6) The called unit sends a **Connect** when it ''answers''.

7) A **Connect** is also sent to the calling unit to let it know the called unit has accepted the setup.

8) The **Connect Acknowledge** message is sent from the calling unit to the switch. The B channel is now ready to pass data.

9) A **Connect Acknowledge** is also sent by the switch to the called unit to complete the message exchange. The B channel is now ready to pass data.

At this point the Bearer Channel has been connected through the network and user data is free to flow. When the call is ready to be disconnected the following exchange takes place.

1) The **Disconnect** message is sent to the switch from the unit requesting the B channel disconnect.

2) A **Disconnect** is also sent from the switch to the far end unit.

3) The switch responds to the **Disconnect** with a **Release**.

4) The far end unit also sends a **Release** in response to the **Disconnect** from the switch.

5) The unit that requested the B channel disconnect will send a **Release Complete** to the switch.

6) The far end unit receives a **Release Complete** from the switch. At this point the call is completely disconnected and the B channel is released and ready for another call.

Although not shown in figures 25 and 26, all Q.931 messages are sent in a LAPD **UI** or **I** frame. If a message is received without errors the acknowledgment response will be a Receiver Ready sent in a LAPD Supervisory frame.

Following the common part of the Q.931 message is the Message-specific part. The Message-specific part is divided into information elements that define the parameters of the message. As an example a setup message would need information elements for the phone number being called, the type of call (voice or data), data rate adaptation to 56Kbps (if needed) etc.

Information elements can be of variable length. When placing calls routers will often use only three information elements following a setup message. Telephones and more sophisticated devices may have more information elements to describe various supplemental services. Q.931 information elements are bit oriented as opposed to character oriented.

First among the information elements used by the setup message is the bearer capability element. This element tells the receiving device and the switching network the basic parameters of the connection requested. The first field is the information transfer capability. The field used in data communications is unrestricted digital information. This selection allows the user to place any type of data on the B channel without the network altering it in any way.

When speech is selected it informs the network that the information on the bearer channel is 64Kbps, PCM encoded voice communications. With this information the network knows that it can decode the signal to analog if the phone number dialed is a Pots line.

3.1 kHz audio is used to inform the network that the information is PCM encoded analog modem signals. This allows a TE to PCM encoded V.32 modem signals and dial-up any standard modem on a Pots circuit. While this sounds like speech there is an important difference. Under some conditions voice calls may be routed through compression devices to increase the number of calls on a trunk circuit. This will cause only a mild degradation in voice quality but would cause problems with hi-speed analog modems. The 3.1 kHz name results from the bandwidth of the audio signal on a standard 2 wire analog phone circuit (300-3400 kHz).

Another important field within the bearer capability element is the User Rate. Q.931 contains protocols for a variety of user rates based upon standard byte or bit repeating rate adaptation schemes. There are also provisions for the data to be ASYNC or SYNC. Many routers use integral ISDN dial backup. These routers often support only full 64Kbps or ITU rate adaptation to 56Kbps. When rate adaptation to 56Kbps is used the router will not use bit position 8 in the 2B1Q frame. This is important because there is the possibility that the Telephone Company will route the call over trunk lines that are not 64Kbps clear channel or the remote may be switched 56Kbps service. General purpose ISDN terminal adapters that connect to a PC generally includes several forms of rate adaptations that allows for both sync and async user data with a wide variety of speeds. When calling TE to TE the call is still 64 Kbps per B channel with these rate adaptations. It is the method of placing the data on the B channel that determines the user data rate.

The second information element is the channel identification field. This field is used to determine which B channel will be used for the call. It is common for the TE to allow the network to determine which channel is used.

Next is the Calling Party Number field. This contains the source address of the location making the call. This can be used for caller ID purposes with voice calls. When ISDN is used for dial backup in data networks this can be used for call blocking and ID purposes. A list of acceptable Calling Party Numbers can be configured in the router and it will ignore calls from other locations. If subaddressing is used the Calling Party subaddress information element will follow the Calling Party Number.

The Called Party field follows with the destination number for the call. This field is used regardless of the numbering plan employed. The number of delivered digits can vary somewhat depending on the country and the numbering plan used. BRI often provides the entire phone number while PRI often limits the presented digits to the last 2 or 4. If subaddressing is used it will immediately follow the Called Party field.

A Keypad field is sometimes seen in the message also. With National ISDN 1 and 2 the Keypad field may be substituted for the Called Party Number but this is not the intended function of the field. The Keypad field is intended to invoke supplemental services and reflects the numbers pressed by the user on the telephone keypad.

Under the Disconnect message there is also an information field. This field, called the Cause element, will contain the reason the call was terminated. Reasons include Normal Call Clearing, Busy, No Answer, etc.

Another aspect of placing ISDN calls is the ability of a TE to reject a call because of incompatibility. An example of this would be if a telephone was used to call a data only TE such as a router backup device. Such a call would have a value of speech within the Setup Message. After the router examined the message it would not return a Connect to the local switch but a release. In this way a device on an ISDN circuit will not answer a call that does not match it’s capabilities. Frequently a device will have more than a single capability and can adapt to the circuit request on a call by call basis based upon the contents of the Setup message. This capability is common in standalone home TAs that support both voice and data.

Figure 27a-i on the following pages show a breakdown of the information transferred between a BRI router backup device and the local telephone switch. This D channel exchange is preceded by a list of Q.931 messages and cause codes. Because switches can be configured in a variety of ways and with different software these messages should be viewed only as general example of the message types and information elements that may be used within the messages. The selection of messages was made using a router backup device configured for the simplest operation possible.

When viewing the message transfers between the switch and an ISDN TE it will be helpful to refer to figure 25. This will show the relationship of the message to the call establishment phase. Upon examination it should be noted that some information elements are associated with only one Q.931 message, while others can be used with a number of different messages. An example of this would be the Channel ID information element. This element is used in the call proceeding message from the switch to the originating TE and in the Alerting message to the answering TE.

In the example the telephone number called by the originating TE is shown in the keypad information element of the Setup message. The ITU specification requires information of this type to be sent in

International Alphabet Type Five or IA5 characters. IA5 is, for the most part, the same as the ASCII code. The switch used in these calls did not present the called or calling party address with is rather unusual.

A complete list of Q.931 messages and information elements can be found in the appendix. Breakdowns of each sub-field of the information elements bit by bit is not possible within the scope of this paper but is available in the ITU Q.931 document.

After successful completion of the call setup the user has a connection end to end over the B channel that was assigned by the switch. During the time that the B channel is in use the TE will still have ongoing communication with the local switch over the D channel. This communication can take the form of a simple poll and response using layer 2 receiver ready frames when there is no layer 3 information to exchange. Switches can also poll a unit with a layer 3 Status message to verify if any calls are active or in progress. A single D channel is capable of controlling more than one call, or B channel, simultaneously.

When the user is ready to disconnect the call a series of Q.931 messages will be sent over the D channel to disconnect the B channel circuit. Terminal Adapters often have more information fields within the setup messages than shown in the following pages. Information in the SETUP message is based upon the capabilities and requirements of the TE.

Message Type	**Hex**	**Binary**
Alerting	01	0000 0001
Call Proceeding	02	0000 0010
Progress	03	0000 0011
Setup	05	0000 0101
Connect	07	0000 0111
Setup Acknowledge	0D	0000 1101
Connect Acknowledge	0F	0000 1111
User Information	20	0010 0000
Disconnect	45	0100 0101
Restart	46	0100 1101
Release	4D	0100 1101
Restart Acknowledge	4E	0100 1110
Release Complete	5A	0101 1010
Cancel	60	0110 0000
Facility	62	0110 0010
Register	64	0110 0010
Facility Acknowledge	6A	0110 1010
Notify	6E	0110 1110
Facility Reject	72	0111 0010
Status Enquiry	75	0111 0101
Congestion Control	79	0111 1001
Information	7B	0111 1011
Status	7D	0111 1101

Codes Set 0	**Hex**	**Binary**
Segmented Message	00	0000 0000
Bearer Capability	04	0000 0100
Cause	08	0000 1000
Connected Number	0C	0000 1100
Call Identity	10	0001 0000
Call State	14	0001 0100
Channel Identifier	18	0001 1000
Facility	1C	0001 1100
Progress Indicator	1E	0001 1110
Network Specific Facilities	20	0010 0000
Terminal Capabilities	24	0010 0100
Notification Indicator	27	0010 0111
Display	28	0010 1000
Date/Time	29	0010 1001
Keypad	2C	0010 1100
Keypad Echo	30	0011 0000
Information Request	32	0011 0010
Signal	34	0011 0100
Switch hook	36	0011 0110
Feature Activation	38	0011 1000
Feature Indication	39	0011 1001
Service Profile ID (SPID)	3A	0011 1010
Endpoint Identifier	3B	0011 1011
Information Rate	40	0100 0000
End to End Transit Delay	42	0100 0010
Packet Layer Binary Parameter	44	0100 0100
Packet Layer Window Size	45	0100 0101
Packet Size	46	0100 0110
Calling Party Number	6C	0110 1100
Calling Party Subaddress	6D	0110 1101

Called Party Number	70	0111 0000
Called Party Subaddress	71	0111 0001
Redirecting Number	74	0111 0100
Redirection Number	76	0111 0110
Transit Network Selection	78	0111 1000
Restart Indicator	79	0111 1001
Lower Layer Compatibility	7C	0111 1100
Higher Layer Compatibility	7D	0111 1101
User to User	7E	0111 1110
Escape for Extension	7F	0111 1111
Transit Delay Selection	83	1000 011

ITU-T Q.931 Cause Codes

ISDN Progress Messages

1	*Not End to End ISDN Call*	Some portion of the network is non-ISDN (56K).
2	*Destination not ISDN*	Destination user equipment is non-ISDN.
3	*Origination not ISDN*	User equipment originating call is non-ISDN.
4	*Call Returned to ISDN*	Non-ISDN network has returned a call to the ISDN network.
8	*In Band Info/Pattern*	The B channel contains an info pattern (i.e. PCM encoded ring back tone).
10	*Unit Could Not Answer*	The Remote was sent a setup but did not respond.
16	*No Response, Re-attempted*	
17	*Treatment applied to Call*	
18	*Call Proceeding*	Normal indication of call being setup by network.
19	*Alerting at destination*	
20	*Connected at Destination*	
21	*Dialing*	
22	*Disconnect*	

ISDN Cause Codes

1	*Unassigned Number*	The Destination number is not a working number.
2	*No Route To Transit Net*	The requested transit network is unrecognized.
3	*No Route to Destination*	You can’t get there from here.
6	*Channel Unacceptable*	Unacceptable B channel requested.
7	*Call Awarded*	Packet mode cause code.
16	*Normal Call Clearing*	Not a problem, normal disconnect.
17	*User Busy*	Do you need an explanation?
18	*No User Responding*	The called user has not responded to the SETUP message.
19	*No Answer from User*	Equivalent to ring no answer.
21	*Call Rejected*	Can be generated by switch or answering unit. May indicate wrong SPID.
22	*Number Changed*	The relationship is over, get on with your life
26	*Selected User Clearing*	The user was not allowed to accept an incoming call.
27	*Destination Out Of Order*	Setup message could not be delivered to remote TE.
28	*Invalid Number Format*	Dialed phone number format invalid.
29	*Facility Rejected*	Facility requested by the user is unavailable.
30	*Response to Status*	Unit responding to network status inquiry (not a problem just informational).
31	*Normal Unspecified*	Used when no other code applies (My favorite, it means we know but we ain’t tellin).
34	*No Circuir/Channel Avail*	Telco has no available circuits (Mothers Day problem).
38	*Network Out Of Order*	Network down (don’t try call for a while).
41	*Temporary Net Failure*	Network down temporarily (try call again now).

42	*Switch Congestion*	High switch traffic, call cannot be handled(Mothers day again).
43	*User Info Discarded*	Low layer information could not be forwarded to the called unit.
44	*Circ/Channel Not Avail*	Circuit channel not available, may be long distance carrier problem.
47	*Resource Unavailable*	General message to indicate the call cannot be processed (Nonspecific).
49	*Quality Not Available*	X.213 quality parameter cannot be supported.
50	*Facility Not Subscribed*	This circuit does not support the requested service (i.e. data call on a voice only circuit). Frequently seen with wrong SPID.
51	*Bearer Incompatible*	Like above but SPID OK.
53	*Service Op Violated*	You must have done something very bad to get this.
54	*Incoming Calls Barred*	You called an answer only circuit.
57	*Capability Unauthorized*	Unit not authorized to use the requested service
58	*Capability Unavailable*	Requested capability not presently available(try later).
63	*Service/Opt. Unavailable*	
65	*Capability Unimplemented*	Requested bearer capability not supported.
66	*Chan. Type Unimplemented*	Requested channel type not supported.
69	*Facility Not Implemented*	Supplemental service not implemented.
70	*Restricted Info Only*	Circuit does not support unrestricted digital info.
79	*Serv. /Opt. Unimplemented*	i.e. don't try ''hold'' if you didn't pay for it.
81	*Invalid Reference Value*	See call reference value in layer 3 Q.931
82	*Channel Does Not Exist*	
83	*ID Does Not Exist*	
84	*Call ID in Use*	
85	*No Call Suspended*	Sort of a hold type function error message.
86	*Call Has Been Cleared*	The call has been disconnected.
88	*Incompatible Destination*	Remote does not support bearrer type.
90	*Direct Call Not Subscr.*	
91	*Invalid Transit Net Sel.*	
95	*Invalid Message*	General Q.931 message problem.
96	*Mandatory IE Missing*	Required Q.931 element missing.
97	*Message Unimplemented*	
98	*Message Not Permissible*	
99	*IE Unimplemented*	An unsupported info elernent was used.
100	*Invalid IE Contents*	A field within an info element is miscoded.
101	*Message incompatible*	A Q.931 in not compatible with the call
102	*Recovery on timer expiry*	Certain Q.921/931 things didn't happen within a prescribed time.
111	*Protocol Error*	A layer 3 (i.e. Nat 1) protocol error has occurred.
118	*Invalid Calling Number*	Your setup message has the wrong LDN.
127	*Cause Unknown*	Odds are high that you will get this one.

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00000101	05	Q.931 message = **Setup**
	00000100	04	information element = **Bearer Capability**
	00000010	02	information element length = 2 octets
	1-------	88	extension bit = information not continued
	-00-----	--	code standard = ITU (standard)
	---01000	--	info transfer = unrestricted digital info
	1-------	90	extension bit = information not continued
	-00-----	--	transfer mode = circuit mode (not X.25)
	---10000	--	information transfer rate = 64 Kbps
	00011000	18	information element = **Channel ID**
Layer 3	00000001	01	information element length = 1 octet
	1-------	83	extension bit = information not continued
	-0------	--	Interface ID = implicitly identified
	--0-----	--	interface type = basic rate
	---0----	--	spare
	----0---	--	preferred/exclusive = preferred
	-----0--	--	D channel indicator = non-D channel
	------11	--	B channel selection = any B channel
	00101100	2C	information element = **Keypad**
	00000111	07	information element length = 7 octets
	00110001	31	IA5 digit = 1
	00110010	32	IA5 digit = 2 phone number
	00110011	33	IA5 digit = 3 123-4567
	00110100	34	IA5 digit = 4
	00110101	35	IA5 digit = 5
	00110110	36	IA5 digit = 6
	00110111	37	IA5 digit = 7
	XXXXXXXX	XX	frame check sequence (octet 1)
Layer 2	XXXXXXXX	XX	frame check sequence (octet 2)
	01111110	7E	end flag

Q.931 setup message form TE to Network

Figure 27a

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00000010	XX	Q.931 message = **Call Proceeding**
	00011000	18	information element = **Channel ID**
	00000001	01	information element length = 1 octet
Layer 3	1-------	19	extension bit = information not continued
	-0------	--	interface ID = implicitly identified
	--0-----	--	interface type = basic rate interface
	---0----	--	spare
	----1---	--	preferred/exclusive = exclusive
	-----0--	--	D channel indicator = non-D channel
	------01	--	info channel selection = channel B1
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Call Proceeding Message from Network to TE
Figure 27b

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
Layer 3	00000001	01	Q.931 message = **Alerting**
	00110100	34	information element = **Signal**
	00000001	01	information element length = 1 octet
	00000001	01	signal value = ring back tone on
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Alerting Message from Network to TE
Figure 27c

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
Layer 3	00000111	07	Q.931 message = **Connect**
	00110100	34	information element = **Signal**
	00000001	01	information element length = 1 octet
	00111111	3F	signal value = Tones off (ring back)
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Connect Message from Network to TE

Figure 27d

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
Layer 3	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00001111	0F	Q.931 message = **Connect Acknowledge**
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Connect Acknowledge Message from TE to Network

Figure 27e

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00000101	05	Q.931 message = **Setup**
	00000100	04	information element = **Bearer Capability**
	00000010	02	information element length = 2 octets
	1-------	88	extension bit = information not continued
	-00-----	--	code standard = ITU (standard)
	---01000	--	info transfer = unrestricted digital info
	1-------	90	extension bit = information not continued
	-00-----	--	transfer mode = circuit mode (not X.25)
	---10000	--	information transfer rate = 64 Kbps
	00011000	18	information element = **Channel ID**
Layer 3	00000001	01	information element length = 1 octet
	1-------	83	extension bit = information not continued
	-0------	--	Interface ID = implicitly identified
	--0-----	--	interface type = basic rate
	---0----	--	spare
	----1---	--	preferred/exclusive = exclusive
	-----0--	--	D channel indicator = non-D channel
	------01	--	B channel selection = use channel B1
	00110100	34	information element = **Signal**
	00000001	01	information element length = 1 octet
	01000000	40	signal value = alerting pattern 0 on (ring)
	XXXXXXXX	XX	frame check sequence (octet 1)
Layer 2	XXXXXXXX	XX	frame check sequence (octet 2)
	01111110	7E	end flag

Q.931 setup message form Network to TE
Figure 27f

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
Layer 3	00000001	01	Q.931 message = **Alerting**
	00011000	18	information element = **Channel ID**
	00000001	01	information element length = 1 octet
	1-------	19	extension bit = not continued
	-0------	--	interface ID = implicitly identified
	--0-----	--	interface type = basic rate interface
	---0----	--	spare bit
	----1---	--	preferred/exclusive = exclusive
	-----0--	--	D channel indicator = non-D channel
	------01	--	information channel selection = channel B1
	00000001	01	signal value = ring back tone on
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Alerting Message from TE to Network

Figure 27g

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
	00000000	00	SAPI address = 0 (call control)
Layer 2	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
	00000000	00	protocol discriminator (ITU Q.931)
Layer 3	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00000111	0F	Q.931 message = **Connect**
	XXXXXXXX	XX	frame check sequence octet 1
Layer 2	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Connect Message from TE to Network

Figure 27h

Layer	Binary	Hex	Decode
	01111110	7E	Start Flag
Layer 2	00000000	00	SAPI address = 0 (call control)
	XXXXXXXX	XX	TEI address (network assigned)
	XXXXXXX0	XX	next send (NS)
	XXXXXXXP	XX	next receive (NR)
Layer 3	00000000	00	protocol discriminator (ITU Q.931)
	00000001	01	length of call reference = 1 octet
	XXXXXXXX	XX	call reference value (random number)
	00001111	0F	Q.931 message = **Connect Acknowledge**
	00110100	34	information element = **Signal**
	00000001	01	information element length = 1 octet
	01001111	4F	signal value = signal off
Layer 2	XXXXXXXX	XX	frame check sequence octet 1
	XXXXXXXX	XX	frame check sequence octet 2
	01111110	7E	end flag

Q.931 Connect Acknowledge Message from Network to TE
Figure 27i

Requirements to answer an ISDN Call

When a call comes into a BRI or PRI ISDN device the following procedure should be used by the device receiving the call to determine what action to take. The process is based upon a series of comparisons, made by the unit, between the incoming setup message and the unit's configuration. In order to answer a call all comparisons must return a result of valid. If any of the results returned is invalid the call should be ignored. While the order listed below follows the order found in the Setup Message this is not a requirement, information elements can be examined in any order and the result should be the same.

When calls are coming into a PRI or U interface BRI TE it is allowable to send a Disconnect in response to a Setup that is invalid. This is because the connections are considered Point to Point and there can be no other devices sharing the circuit. This is not the case with S interface BRI. Because of the multipoint nature of the S interface a call that may be invalid for one unit may be valid for another. With each TE having knowledge of only it's configuration, and not the configuration of other TE's sharing the same circuit, it is not in a position to make a judgment on what constitutes a valid call for another device on the interface. On an S interface the only option is to ignore a Setup that is invalid.

Bearer Capability Information Element Information Transfer capability

While there are a number of possible coding options for this element there are three that are the most common. *Unrestricted digital information* represents standard data calls. This could be raw data or data encoded in V.120, V.110, PPP. Other information elements may be presented to inform the answering unit which of these encapsulation methods is employed.

Speech and *3.1 KHz Audio* can be treated the same by the answering device. The difference between them has more to do with the calling equipment. From the perspective of the answering unit the data stream presented is G.711 PCM encoded analog signals. Bear in mind that this can be either Mu-law or A-law. When calling between countries that use different PCM encoding it is the responsibility of the Telephone Company in the Mu-law country to do the conversion.

Information Element User Rate

The *User Rate* part of the bearer capability IE list the actual data transfer rate. Generally this will be either 64 or 56 Kbps. While operation at 56K is not an issue in Europe, Asia or Latin America it is very important in North America. If this information element is present it will define the call as V.110 56K, in the case of rate adapted data calls. If this information element is not present then the default of 64K is assumed. Optionally the unit could be designed to reconfigure itself based upon the contents of this information element. If this information element is present in the Setup message it must match the units configuration or the call is considered invalid.

Called Party Number Information Element

In North America this information element will always present the 7 digit local directory number (LDN) for BRI and, usually, the last 4 digits for PRI. In Net 3 countries the number of digits presented can vary widely. As little as 1 and as many as 4 digits are commonly seen with or without multiple subscriber numbers (MSN). Because of this it is important that number checking start at the least significant digit and proceed until either the local directory number, entered in the unit, or the called party information element of the Setup message runs out of digits. Examples are given in figure 28 below of both valid and invalid called party setup numbers.

Support for a single phone number, assigned to both B channels, is supported in Net 3 countries. In this case the user may wish to leave the LDN (MSN) field blank in the units configuration. If the field is left blank the call is placed in contention and all devices may compete to answer it. Contention resolution will most likely be based upon a priority system in a single TE. In an S interface environment the contention resolution of the S interface will decide the winner.

LDN in TE	Called Party Number in Setup	Result
4721234	34	Valid
4721234	4721234	Valid
4721234	3104721234	Valid
1234	34	Valid
1234	4721234	Valid
1234	3104721234	Valid
472	34	Invalid
472	4721234	Invalid
472	3104721234	Invalid
‘’blank’’	‘’any number’’	Valid
‘’any number’’	‘’blank’’	Valid

Called Party Number validity chart

Figure 28

Called Party Subaddress Information Element

Subaddressing is sometimes used in NET 3 (EURO ISDN) countries as way to direct data calls to a particular device on an ISDN circuit. In some locations in South America it is the only supported numbering plan. Because the subaddress is user defined, and passed blindly by the network, call acceptance should require an exact match of the subaddress. If the subaddress entered into the answering unit differs from the called party subaddress IE the call should be considered invalid. If the called party subaddress field in the TE’s configuration is blank then this field should be ignored and the call considered valid without regard to the presence or absence of subaddress information element in the call setup message. Subaddressing should support IA5 alphanumeric characters.

Calling Party Number Information Element

Calling party numbers can sometimes be used to identify the location placing the call. This can be used as a form of security to only allow calls from certain locations. To make use of this feature the TE must have a table where valid calling party numbers can be stored for comparison to incoming Setup messages. Due to tariffs, legal restrictions, and international incompatibilities, this information element cannot be relied up to be present. In order to use it the circuit must be tested end to end to verify the presence and contents of this Information element.

In National ISDN 1 switches this information field is often not present and the calling party number is placed, instead, in another information element. This is the display text IE that is intended to place alpha numeric information on an LCD screen built into the ISDN terminal adapter or used by Computer Telephony Integration (CTI) systems. When this field is present but the calling party number is not available (call from PBX, analog call from old switch etc., circuit limited to 56kbps) the Display Text field will be coded as ''Unknown''.

If the decision is made to use the calling party number, to restrict incoming calls to those from certain locations, then the matching procedure should be the same as for the called party number above. If the feature is enabled in the unit, but no calling party number is delivered in the setup message the call should be considered invalid.

Calling Party Subaddress Information Element

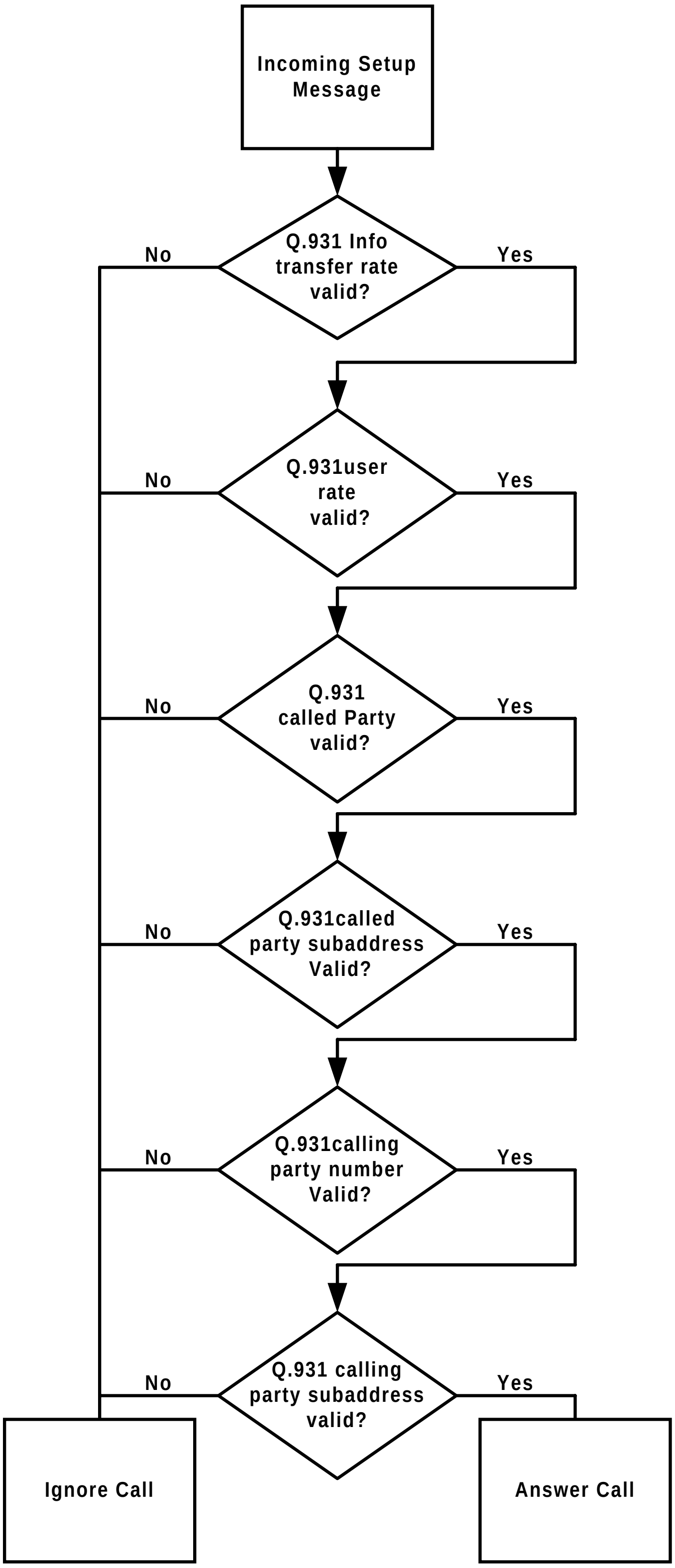
As with the called party subaddress above this will only be found in NET 3 countries. Again since it is user defined and passed blindly by the network an exact match should be required. If no calling party subaddress field is offered in the answering TE, or if the field is left blank, the call should be considered valid.

Lower Layer Capability Information Elements

When present this Information Element contains details such as V.120 rate adaption information, user interface speed etc. While conceptually there is some merit to the concept this IE is not always passed through the network. Because of this it cannot be relied upon to be present, however, if it is present the information will be accurate. Experience has shown that it is best, for the most part, to ignore this IE.

Call Validation Flow Chart

When a setup message is delivered to ISDN terminal equipment a series of validation checks should be performed to determine if the call should be answered. If the result is positive for each of the checks the call should be answered. If any of the validations results in a negative result then the call should be ignored. In most cases BRI terminal equipment is unaware of how many if any other devices are sharing the same circuit. For this reason calls are generally not rejected but ignored if they do no pass the validation checks. A call number validation flow chart is shown in figure 29.



Call Validation Flow Chart
Figure 29

Subaddressing, MSN and LDN

In an effort to clear up some confusion between NET 3 subaddressing, Multiple Subscriber Numbers (MSN) and National-ISDN Local Directory numbers (LDN) the following is a brief explanation of each method and highlights the differences. Central office telephone switches can be configured in a number of ways so that the information that follows may vary for application to application.

In NET 3 the standard circuit for a BRI terminal adapter will have one telephone number for both B channels. If the user needs to place a call to a specific ports on a multiport TA they may use subaddressing. In this method the remote user wishing to place a call to port 2 of the TA would dial the circuit phone number plus the subaddress assigned by the user to port 2. Although methods differ one way to perform this is with an identifier in the AT command string. For one BRI terminal adapter the command ATDT4245555#2 will generate a setup message with a called party number of 424-5555 and a called party subaddress of 2. The actual command may vary between manufacturers.

The setup message received by the answering unit will contain both a called party number field of 4245555 and a called party subaddress field of 2. Remember that the entire local phone number may not be included in the called party field at the answering site. The number of digits presented is country dependent. It is not uncommon for the local Telephone Company to only deliver 1, 2, 4, or all digits to the answering unit. The TE equipment allows the user to pick their own numbers as subaddresses assigned to the various ports. The Phone Company does not assign subaddresses it just delivers the alpha numeric subaddress character generated by the calling TE.

In the event a call Setup message with no subaddress field comes in, the TE may still allow the call to be answered, however, there is no way to direct the call to a specific port. It is up the TE manufacturer to determine how to deal with these situations. Generally when a subaddress is configured in a TE but no subaddress is delivered the lowest available compatible port will answer the call.

Subaddressing is primarily used for data calls. This is because users cannot dial subaddresses on conventional analog telephones. Subaddressing solves the problem of directing calls to individual ports on a TA, or devices on the S/T bus without assigning numerious phone numbers. Because of the inability of all calling devices to utilize the subaddressing format it tends to be unreliable for voice calls.

To solve these problems the user may choose to have the MSN feature enabled. In this mode the Telephone Company will assign a single telephone numbers to the circuit. Within the assigned phone number the last digit is not fixed but is a *Wild Card*. For example if the Telephone Company assigns the number 424-555**0** to the circuit, the last digit, 0, is actually a wild card. As a result the actual phone number of the circuit is 424-555X where X can be any digit between 0 and 9.

Using this example a user could configure the TE data port to answer 424-5550, voice 1 will answer 424-5551 and voice 2 will answer 424-5552. On an S/T interface this could be carried to other TAs. The phone number presented to the answering TE by the central office switch may be l, 2, 4, or all the digits of the called party number. The TE should start checking the MSN from the least significant digit. As a result only 1 digit of the MSN number needs be assigned to a port on the TE to achieve call direction in the example given.

Because telephone companies in some countries are concerned about running out of telephone numbers MSN can be a very expensive option or not available at all. Some Telephone Companies will divide the MSN between users giving the digits 0-4 to one user and 5-9 to another user. This helps preserve valuable telephone numbers and reduces cost to the user.

Popularity of one Subaddressing over MSN in a particular application is dependent upon the cost of the MSN option and the intended use of the ISDN circuit. Remember POTS phones can call numbers directly on an MSN enabled circuit, which is why it is popular in many applications.

One other method of numbering in NET 3 is B channel numbering. This method assigned a separate phone number to each B channel. This means that with a two port TE all calls to and from port 1 would use B channel one. All calls to and from port 2 would use B channel 2. This has fallen out of favor and is seen less and less with the advent of MSN and subaddressing.

If none of these methods are used a single phone number without subaddressing or MSN can be applied to the circuit. In the event the TE has more than one port the call in considered in contention and is answered by first port to respond.

In North America the approach to numbering plans has taken a different road. Local Directory Numbers (LDN) are used in the US. With some of the older switch software it was possible to only have one phone number assigned to a 2B+D circuit however this has fallen by the wayside with the implementation of the National ISDN-1 standard. As it now stands if a user wants a circuit with both B channels active they will have a minimum of 2 LDNs assigned by the Phone Company. Unlike NET 3 environments there is no subaddressing. Multiple LDNs is not a special order feature but the common method of call direction. LDNs are ABSOLUTE there is no wild card or user defined numbers.

Most central office switches can support a minimum of 10 telephone numbers per ISDN circuit depending upon the CO switch. The most common circuit configuration, and those recommended for most BRI terminal equipment, will have 2 LDNs. These phone numbers do not have to be consecutive and it is common for the last 4 digits to be quite different. Unlike MSNs were the last digit is used by the TAs determine how to route an incoming call to a port, the entire 7 digit LDN is examined by the TE on an incoming National-1 call for the purpose of directing the call to a voice or data port.

When configuring BRI terminal equipment for use on National-1 circuits the user will enter a SPID that matches each LDN. Each SPID will be assigned to a unique TEI (64-126) and use that TEI to place and answer all calls associated with that SPID. Within the SPID can be found the LDN. As a result there is a relationship between a TEI the SPID the LDN and the PORT to which that LDN is assigned. Note that the B channel is not included in this relationship. B channels can be dynamically assigned on a call by call basis.

The normal method of configuration for BIR equipment is to use one SPID / LDN for the Data port and use the same one again on one of the voice ports. The other SPID / LDN will be used only for the remaining voice port. In this way if a call comes in for the data / voice LDN it will be answered based upon its bearer capability being DATA or SPEECH. When a call comes into the voice only LDN the port will ring only if the call is a SPEECH or 3.1Khz call. In the event a data call is active and a voice call is placed to that same number the switch may consider the number busy. This inability to have two calls active against one LDN is switch configuration dependent but commonly supported.

It is possible to order a BRI circuit with 3 LDNs. In this mode the data port and each of the two voice ports have unique telephone numbers. While this can be very convenient it does come at additional cost. The logic behind the 2 LDN method is that you can never have more that two active calls at any given time not including calls on hold. This should allow 2 LDNs to be sufficient.

As you can see while the purpose of subaddressing, MSN, and LDN is to direct calls to a specific port of a specific TE on a given Basic Rate circuit the methods employed to achieve this are quite different. While it is common to hear that MSN is the European (NET 3) name for our LDN the truth is that while similarities exist they are quite different in many way and should not be considered interchangeable. A list of features for MSN, LDN and subaddressing is shown in figure 30.

Subaddressing is sometimes available on a special order basis in the US, however, it is very rarely used in North America and most TAs sold here do not support it. Below is a summation of the differences between MSN and LDN. Be aware that some of the National ISDN features vary depending upon the switch used at the serving CO.

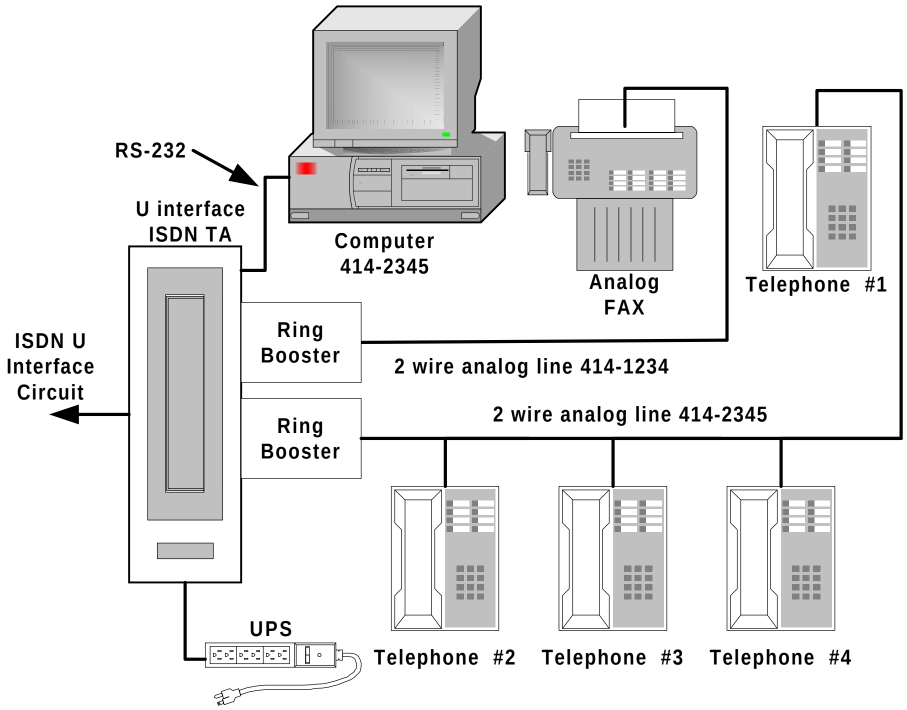
Feature	MSN NET 3	LDN National-1	SA
Max telephone numbers per circuit	9 (0-9)	Up to 64 (switch dependent)	Over 100
Consecutively numbered	Yes	No	No
Number of TEI assigned	1 per TE	1 per LDN / SPID	1 per TE
Multiple calls per TEI	Yes	Yes (switch dependent)	Yes
Important digits in MSN	LSD	All 7 digits	LSD
Optional or standard feature	Option	Standard	Option

Comparison of MSN - LDN - SA
Figure 30

In the following drawing the ISDN TA is a type manufactured by a number of companies. Following is a basic list of features for operation in North America.

- U interface connection to the telephone company.
- 1 RS232 data connection to the PCs' com port.
- 2 analog ports that operate with standard telephones.
- The unit should support 2 SPIDs and 2 phone numbers.
- The unit may support call hold, call forward and call conferencing.
- The unit should support the Bellcore EZ1 ISDN configuration package.

The UPS (**U**ninterruptible **P**ower **S**upply) is important because if power is lost none of the telephones would work without it. One manufacturer did produce a small TA with batteries inside that allowed for limited telephone operation during power failures.



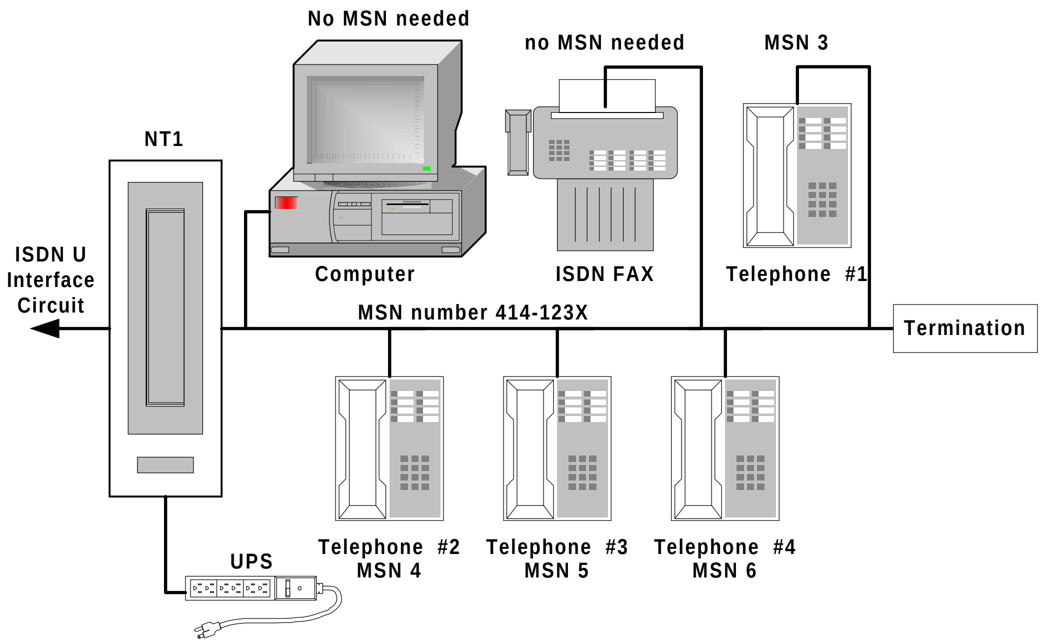
U interface installation
Figure 31

In figure 31 the Ring Boosters (check Vikings Electronics on the WEB) provides enough power to ring all of the telephones. This is not required if the TA ring output is enough to support all of the devices attached to the analog ports but the ability to ring more than one or two phones is often a problem. Add one to many devices and nothing rings.

The fax will operate with phone number 414-1234 while telephones 1, 2, 3 and 4 operate with number 414-2345. It is also possible to use two line phones and let the user select the line to use. Telephone number 1, for example could use the same line as the fax machine with all of the typical problems that entails.

The analog phones and fax machine use classic DTMF for communication to the TA. It is the job of the TA to convert the tones produced by the telephones into D channel messages. The TA must also convert an incoming setup message into ring voltage for the analog lines.

There should be no problem with using both B channels with the PC for multi-link PPP connections but one of the B channels would need to be dropped from the data call to place or answer a telephone call. Remember that most Terminal Adapters have high-speed data ports (230 KBPS) so think about that when you get a PC. With the exception of the ISDN TA all of the equipment, answering machine, telephone, and fax are off the shelf department store items.



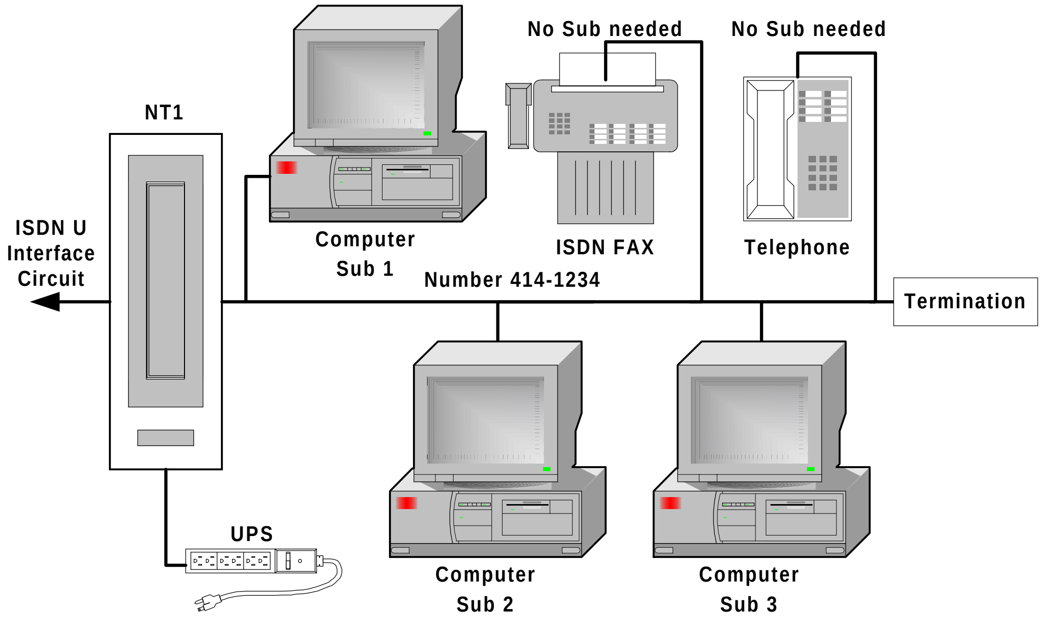
S/T Installation MSN
Figure 32

Figure 32 shows a similar setup with the S/T interface and MSN. While a perfectly valid approach it does have drawbacks. Remember that each device connected to the S/T interface must have an ISDN TA built into it. ISDN telephones are not cheap. ISDN fax machines are also hard to find and the network shown would require 3 pairs of wires. tx, rx and power, to each device. Termination resistors will either need to be added to the end of the circuit or built into the last device on the circuit which is telephone number 1 in the figure.

The NT-l is used to convert the U interface into an *S/T* interface. All of the devices on the circuit are fully ISDN compatible. This means that each one can communicate over the D channel. In North American applications each device would need to have at least one SPID. Remember that in the standard Bellcore IOC configuration packages, for home use, there only two SPIDs. This means that only one or two devices could be attached to the S/T interface. In ETSI countries that do not use SPID numbers this is not a problem.

For a North American installation the user would need at least 5 SPIDs. In this network the Phone Company would need to setup a more complex configuration that allows calls to be directed to a device based upon central office switch programmed criteria. In essence it allow a call to be directed to a device based upon its SPID using the USID and TID fields. This allows each unit to have a unique SPID and the call to be directed to a device or multiple devices based upon the SPID. In the event the phone is not answered the CO switch could roll the call over to the next device in its table. This type of installation is primarily for voice related business purposes.

As you can see this type of installation results in increased cost with little, if any, benefit for the average home user. It is also far too complex for the average dial backup application for routers and the like.



ETSI installation
Figure 33

For applications that are primarily data in nature the method in figure 33 can work well when used with subaddressing in an ETSI network. By providing a unique subaddress to each PC the user can place a call to any PC. The telephone will answer all calls with a bearer capability of Speech and does not need a subaddress.

The fax machine will answer all calls defined as ISDN FAX in the setup message. This presents the problem of fax compatibility. Only ISDN fax machines can communicate with this type of fax machine.

When ordering a PRI circuit the user may have two choices of numbering plans. It is most common to get a circuit with one phone number for all 23 B channels. When this is selected the B channel on which the call is placed is negotiable between the switch and the CPE. Two methods are commonly employed to determine which B channel is to be used for an incoming or outgoing call.

- Use the lowest number available B channel. As calls are disconnected on low number B channels they become the next one used. With this method higher number B channels such as channel 23 can be used very seldom.

- Cycle the calls through the B channels. For example the first call will go on B-l, The second will go on B-2, third on B-3 etc. This will happen even if no other calls are active on the PRI circuit.

If the CPE is capable of directing calls based upon the content of information elements within the setup message then this single number plan may be the preferred method. Most Internet Service providers select this method allowing both Data (ISDN TA) and voice (modem) calls to dial the same number. Incoming calls will be dealt with differently based on the Bearer Capability Information Element.

A second and substantially more expensive numbering option is to assign separate telephone numbers to every B channel. In this configuration incoming calls will be placed on the B channel based on the Called party number. This method of operation is called DDI or direct dial in.

While Calling Party numbers are presented in the Setup message on PRI circuits the called party field is frequently limited to the last 4 digits of the number dialed. If the DDI option is selected every phone number on a PRI circuit will have the same prefix so this does not present a problem.

As with North American PRI, 30B+D PRI can have different numbering plans. Use of subaddressing and MSN are not prohibited but are not commonly employed. Specific numbering options vary from country to country but basically you can have one phone number for all 30 B channels or 30 phone numbers, one for each channel. Called and Caller number presentation will differ between countries but the called party number may not be complete and only display a limited number of digits.

SPID Numbers

Within National 1 there is the need for the TE to download to the local switch its unique SPID or **Service Profile Identifier**. This is a number typically between 7 and 14 digits assigned by the local Telephone Company to each phone number on a Basic Rate Line. When the line is ordered phone numbers and SPID numbers are assigned. The SPID must be entered into each TE (i.e. from the front panel, via AT commands, etc). The SPID is down loaded to the switch, in an information frame, immediately after the SABME and is acknowledged with an information frame from the switch that contains two fields called the USID and the TID.

The SPID number is used by the switch to associate, in its database, a given units attributes, options and capabilities to its assigned TEI address. In this way the switch knows, for example, that it assigned TEI 70 to a unit with the SPID of 30538921210101. When the switch looks into its database it sees that that SPID was supposed to be entered into a TE with a defined set of User Service profiles. These profiles are in effect the network services and options that the customer ordered. Examples of such options are circuit switched data, circuit switched voice, X.25 etc.. The second field is the TID or Terminal ID. This is a number that is unique to each terminal. Together the USID and TID make up the **Endpoint Identifier.**

The plan behind the SPID is that the switch can address units based on a combination of the USID and TID. For example it could address a message to all units with USID 10, or to all units except the one with TID 10. By using the TID and USID the plan was to increase the flexibility of the ISDN circuit. The reality is that many if not most users never uses this flexibility and what looked good on paper is a nightmare in practice. Wrong SPID numbers entered into TEs have been the single biggest causes of problems with ISDN installations. While there is a standard SPID format for National ISDN it is, in practice, not always used by phone companies. SPID numbers are not forgiving they are right or wrong and if you have the wrong SPID number in your TE it will not be able to place or answer any calls. Anyone who has been plagued with SPID problems will agree that god will punish someone for the SPID.

North American SPID format YYYXXXXXXX0101 (area code + local seven digit number + 0101)

A more recent development related to the SPID number is the automatic SPID function. This works by the TE sending a message to the network asking for a list of valid SPID numbers. When the switch returns the list the users TE or attached PC will display all of the valid possibilities. The user then must select the proper SPID from the list. While this is some improvement the fact remains that for the vast majority of Basic Rate ISDN applications the SPID provides no useful function, if it did the rest of the world would use them, and they don't.

In the event a TE contains the wrong SPID number the switch will return an error message with a cause code of *Invalid Information Element Contents*. The Source of the cause code will be listed as *Switch Serving Local User*. This response will come immediately after the SPID is sent to the switch.

Another symptom of an incorrect SPID is the inability to place or answer calls. Any attempt do so will result in a *Facility Not Subscribed* message being returned by the switch. This is because without a proper SPID being your TE is not authorized to place or answer any type of connection be it speech or data.

BONDING B1 and B2 for 128 Kbps

BONDING is defined under a paper published by the Bandwidth ON Demand INteroperability Group, which is a consortium of ISDN equipment manufactures. Within the specification there are four modes of operation numbered 0 through 3. Only mode 1 support is mandatory and often the only mode supported by Standalone BRI TAs. Listed below are the general steps that two units must go through to place or receive a BONDING call.

Both units' musts be configured for BONDING. Calls limited to 56 Kbps per B channel, due to the telephone circuits, allow a maximum speed of 112 Kbps.

The originating unit will place a call to the answering unit over B1. Upon connecting the originating unit will generate an Information Message sent in a special BONDING Information Channel frame over the B1 connection.

When the Answering unit detects Information Channel Alignment it will examine the parameters of the call and return, if the parameters are acceptable, the same parameters that it received in an Information Channel Frame.

The calling unit will now request the phone number to be dialed for the second B channel. Upon receipt of the request the answering unit will look into it's Local Directory Number field, for B2, and transmit the number to the calling unit. The user programs the LDN in when they first install the unit. The calling unit will use that number to generate a D channel setup message, for B2, to the remote.

After the second call is answered the units will use Multi-framing, now present on both B channels, to perform Delay Equalization thus aligning the B channels. Once this process is complete the BONDING Multi-frame is stopped and the units can transfer data utilizing the full available bandwidth of both B channels.

Because the frame alignment procedure happens only once, and is not constantly monitored, any change in the Delay of either B channel may render the connection incapable of error free data transfer. The BONDING specification does not deal with recovery from such a condition, thus leaving it up to the TE manufacturer. Recovery is most commonly done by disconnecting and reestablishing both B channels.

Local directory numbers of answering units are only 7 digits long while BONDING calls may be setup over long distance connections requiring area codes. To compensate for the incomplete phone number sent by the answering unit to the calling unit, the calling unit knows to use any digits above the first seven that were required to place the first call.

Mode 0 is similar to mode 1 in that the exchange of parameters and phone numbers takes place, however, no attempt at delay equalization is made. In this method a separate connection (i.e. V.35) would be required for each B channel to the DTE. It is the responsibility of the DTE to perform the inverse multiplexer and frame alignment functions.

Mode 2 does the same parameter negotiation but utilizes framing to maintain delay equalization and provide auto recovery when channel delay alignment is lost. The overhead needed by the framing is 1/64 of the total bandwidth (1 Kbps from each B channel at 64K). The available user bandwidth is 63/64 of the total bandwidth.

Mode 3 provides the same monitoring function as above but allows the user data rate to be increased in multiples of 8 Kbps. Bandwidth is added as needed. You may have to place a 64Kbps call to add 8Kbps of bandwidth.

X.25 on the D Channel

Transmitting X.25 information over the D channel is done in accordance with the ITU-T X.31 which defines X.25 LAPB to LAPD conversion. In essence a LAPB packet is translated into a LAPD packet. A virtual call is set up between the calling and called ends using the X.25 packet facility of the telephone network. Packets are then transferred between users. When the exchange of information is complete the connection is cleared. This has advantages in point of sale and Automatic teller applications among others. Customers pay a low monthly cost for the ISDN circuit plus a small per packet charge. Connection times for packet switching virtual calls are very fast. Local calls can frequently connect and pass data in less than one second.

Using the D channel for packet data does not prevent it from being used to control circuit switched connections on the B channels, however, Q.931 call control messages will always take priority over X.25

user packet data. A BRI circuit with X.25 over the D channel will have a separate address for the D channel (phone number) and a separate SPID assigned for the X.25 portion of the unit. The TE will have a unique TEI used for the X.25 control functions and will send LAPD X.25 call control and information frames with a SAPI of 16.

When D channel Packet Mode is being used all signaling messages take priority over X.25 data. On a basic Rate line the link rate is 16K but because of contention for the channel the effective throughput will depend upon D Channel availability. Normally discussions of X.25 over the D channel refer to 9.6 KBPS as the operating speed.

X.25 on the B Channel

The current method for X.25 over the B channel is not very dynamic. When this service is ordered the ISDN service provider will build one of the B channels as a permanent connection to the X.25 network. While this is referred to as semi-permanent though it looks permanent to me. No TEI is requested over the D channel and no SAPI is used. The net result is that this mode operates like a standard DDS leased line circuit. When a B channel is allocated for X.25 in National-1 it cannot be used for anything else.

National ISDN 2 changes the above operation quite a bit. In National-2 the TE must use a unique TEI in it's request for the B channel X.25 connection to the packet network. After sending the X.25 Setup and receiving a Connect the X.25 path is established. In order to send packets an X.25 SABM must be sent to the X.25 network, over the now established B channel, and acknowledged with a UA frame. At the completion of this ''handshake'' the TE is connected to the X.25 packet network and may send and receive X.25 packets.

Rate Adaption

V.120

V.120 rate adaption is an option for transferring data at subrate speeds over 64 or 56 KBPS connections. ITU V.120 utilizes a method of formatting data into frames for transmission and removing the framing at the receiving end to restore the original data. V.120 is actually a form of statistical multiplexing. Within the specification there are three classes of operation all of which are based on the Q.922 LAPD protocol used in frame relay (Q.922 is based on Q.921).

- Asynchronous protocol sensitive mode (start / stop)
- Synchronous protocol sensitive mode (user data must be HDLC format)
- Synchronous Bit transparent mode (no format requirement)

In the Asynchronous protocol sensitive mode the TE will strip the start and stop bits from the user data and format it into V.120 frames for transmission. V.120 Frames will also contain an address field, information field, start and end flags, and a frame check sequence. The receiving TE will restore the user data to its original form. Although the term protocol sensitive is used it could be more correctly stated that the mode is format sensitive, that is the number of data bits, parity type, and stop bits.

Synchronous Protocol sensitive mode places the address, control, and information fields, of the user HDLC frame, into V.120 frames for transmission. Frames will contain an address field, information field, start and end flags, and a frame check sequence. The receiving end must remove the V.120 fields and restore the HDLC flags and fcs to the frame before passing it to the DTE.

Bit transparent mode places all synchronous data, from first bit to last, into frames and transmits it. No attempt is made to monitor for errors or overruns. It is up to the DTE equipment to perform integrity checks of the data.

Two modes of operation are allowed for the 3 classes of V.120 operation:

- Multi-frame acknowledged information mode (MF)
- UI-frame-only mode (UI).

In multi-frame mode, after a call is connected, the calling TE will send a SABME. Upon receipt the answering unit will send a UA. Once this handshake is completed the units will transfer data using numbered I frames. Acknowledgment of these frames is required and is usually done with RR frames. Each frame is checked for it's sequence number and FCS thereby insuring error free delivery. Other frames such as RNR and XID may also be supported.

The minimum requirement in the Unacknowledged mode is for the user data to be placed into Q.922 UI frames. An optional XID handshake, prior to user data transfer, may be implemented. Interoperability between ISDN Terminal adapters of from different manufactures depends upon the method of V.120 implementation. Some TAs supports both MF and UI modes. If the user application does it's own error checking, like PPP, then UI mode will provide good results without the added complexity of error checking. Generally Async applications function better with MF mode.

It is possible to place V.120 rate adaption parameters in a Q.931 setup message sent over the Basic Rate ISDN D channel at the time the call is initiated. Ideally the answering unit could use this information to automatically configure itself for the incoming call, however, telephone switches may or may not pass this information on to the answering unit. This hit-or-miss approach to the Setup message contents makes it impossible to exploit what could otherwise be a very useful feature.

V.110 Rate Adaption

Although the V.110 specification covers rate adaption for a number of different speeds its complexity and limitations have made it less popular with newer equipment than V.120. The concept of V.110 is based on building a TDM frame for transmission over the B channel. Depending upon the bit positions used in the V.110 frame and how often the frame repeats a range of speeds can be achieved. Unlike V.110 it is important that both the calling and answering TE be configured for the same V.110 parameters with respect to data format and speed. Due to the lack of flexibility and configuration issues it is best to avoid V.110 rate adaption.

Support for V.110 is often available for 56 Kbps data rates over a 64 Kbps B channel. This is especially important in the US where there are still some trunks that cannot support 64Kbps data rates. The primary reasons for using V.110 is due to in-band signaling that rides in a DS0 along with the user data by a method called rob bit signaling. In these situations the trunk lines exist between switches that are not fully SS7 compliant or the remote device being called is a Switched 56 device.

V.110 at the 56 KBPS rate is very simple to implement. The ISDN basic rate frame between the terminal adapter and the telco switch places the user data into 8 bit bytes with numbered positions. V.110 takes the LSB position and stuffs it with a 1 for transmission using the other 7 bits for user data. Upon receiving a frame the unit knows to only process the 7 MSBs and ignore the LSB.

When a call is placed, using V.110, there is an additional information element in the D channel setup message that informs the Telco switching equipment that V.110 at 56 Kbps will be used. With this information the switching equipment knows that it can drop the LSB position, leaving a 56 Kbps data stream to transport over the trunks. The TE also knows which bit position to ignore on received data. When the remote unit is a switched 56 device the telco switch, serving that unit, knows to remove or stuff the LSB position between the 64 KBPS DSO and the switched 56 circuits. Compatibility between other ISDN TE vendors is generally not a problem.

T- Link Rate Adaption

Northern Telecom developed the T-Link rate adaption method for their 2 wire switched 56 service. The specification provides for a wide range of speeds, however, Most ISDN TAs that support T-Link only do so at 56Kbps. At the 56Kbps T-Link, from the ISDN viewpoint, operates the same as 56Kbps V.110.

2 wire switch 56 service is no longer common, in fact both 2 and 4 wire switched 56 have been displaced by ISDN service except in those parts of North America where the Central Office Switch does not yet support ISDN. Switched 56 service is inflexable and expensive and lacks the universal interoperability of ISDN and as such is due to the inflexibility of the service and customer demand for ISDN.

Point to Point Protocol (PPP)

Many ISDN terminal adapters provide an option specifically for use with Asynchronous PPP. This includes anyone that is dialing up to an Internet service provider and most telecommuters. When this option is selected the TA becomes ‘’protocol aware’’. Incoming PPP packets are known by the 7Eh flag character that frames all PPP packets. As the packet comes into the TA from the PC the start and stop bits are stripped leaving only 8 of the original 10 bits. The packet is then transmitted in a synchronous mode over the B channel. Always try to use hardware flow control with Async PPP and a character format of 8N1.

There is a method within the RFC specification for Async PPP that describes how Async control characters should be sent. It requires that all of these special characters, such as Xon and Xoff, be preceded with an escape sequence character. Taking this requirement to an absurd extreme the PPP packet size transmitted over the ISDN link could be doubled in size due to the escape characters. Of course this never happens.

One of the big advantages to this mode of operation is the throughput that can be obtained. Because an external TA only needs to transmit 8 bits for every 10 bits that come from the PC the throughput, as viewed at the PC com port, with start and stop bits counted, is about 80KBPS without compression over one 64KBPS B channel connection.

Multi-link PPP & BACP

Multi-link PPP has replaced BONDING in many data applications and is supported in almost all TAs. Many ISDN applications, such as the Internet, run user data in PPP packets. With normal BONDING the packet enters the TA and is split bit by bit between the two B channels. On the receiving end the bits are shifted from two streams back to one and it is assumed, by the TA, that everything is OK. Multi-link PPP is a much more dynamic method of inverse multiplexing PPP over multiple channels.

In Multi-link PPP (MP) the PPP packet enters a TA that is aware of the PPP protocol. The TA takes each PPP packet and adds an MP header containing a sequence number. The MP packets are then sent over the first available B channel. The PPP packets may be split into smaller packets when the MP header is attached. Using the sequence numbers the TA at the other end strips the MP information and reassembles the PPP packets into the proper order. In the event of a missing sequence the remote TA will discard all fragments associated with that MP packet. Negotiation happens in the LCP option negotiation phase of the PPP link setup. MP can be used with non-ISDN technologies such as modems and dedicated circuits because the specification is not ISDN dependent. There is also no two-channel limitation imposed by the specification.

Currently if the user has MP running over two B channels and one of the B channels hangs up the other B channel may drop the call as well. There is a protocol called BACP for Bandwidth Allocation Control Protocol. The purpose of BACP is to allow for the adding and removing of MP links on demand. BACP is a true bandwidth on demand standard that will allow a host of applications to add and remove connections as required to adjust for load or B channel allocation.

In a home user environment with ISDN there will be two B channels in use for the Multi-link PPP connection. If the location being dialed is long distance or the service provider bills by time, the user will get a bill for 2 calls to that site. With an External TA connected to a COM port on a normal PC the maximum speed is 115.2 KBPS. Like the PPP mode of many TAs mentioned above MP does Async to Sync conversion so that over two B channels the theoretical throughput is about 158KBPS when viewed at the COM port with start and stop bits. Users should set the com port speed above 115.2 Kbps if possible or throughput may suffer.

Using an internal TE that connects directly to the PC bus can also improve throughput. In this mode many TEs look to the PC as if there are two COM ports attached. These COM ports can be considered as virtual devices because they don't exist as DB9 connectors that can be used by other equipment. Because each COM port can be configured to operate at 115.2 the PC is effectively connected at 230.4 KBPS. This results in maximum use of the ISDN 2B bandwidth.

Microsoft offers an ISDN accelerator pack that contains the MP software that works with two COM ports. This allows the TE to place two calls and the Windows 95 operating system takes care of dealing with framing and switching the MP packets between the COM ports of the PC. This should allow for much wider use of the MP protocol due to a reduction in vendor induced variables to the operation of the MP protocol.

Remember that the Central Site you are dialing into with MP will have to allocate two B channels for you. Because of this there is usually an added cost for the service. Service providers should be aware that if they give all of their customers MP capabilities they will need to double the number of incoming B channels with the resulting increase in telephone line cost and equipment cost. Before agreeing to jump on the MP wagon it might be a good idea to make sure the extra bandwidth is really needed and the extra cost is justified. If you do choose to use it you will find that it provides a tremendous improvement over single B ISDN or V.90 modem connections.

Choosing V.110, V.120, PPP or MP

When installing Basic rate equipment that will operate in Async or subrate Sync modes a choice of rate adaption must be made. Some basic rules can be used to determine what is best for a given user or application.

V.110 should only be used if the location to which you are placing a call mandates its use. In North America it will probably never be required. In NET 3 environments such as Europe and Latin America there is a large installed base of equipment that only supports V.110. If you are in those areas of the world, or calling to them, V.110 may be required.

V.120 is the default method of rate adaption for most North American ISDN equipment. It is protocol transparent so it can be used for dumb ASCII, PPP, HDLC, SDLC and so on. When in doubt, or not specifically told to use something else, pick V.120. It is also displacing V.110 in the rest of the world.

PPP will work very well if you are dialing into a device that never converts the data from asynchronous to Synchronous. For example Windows dialup networking offers the user to select PPP communications. When this option is selected the data leaving the PC Com port will be in Asynchronous PPP format. The ISDN terminal adapter will then convert the data to synchronous before transmission. It is desirable that this data not be converted back to Asynchronous format by the answering device. Most standard ISP dialup access devices will maintain the synchronous format. If the answering device is going to convert back to asynchronous format the user should consider using V.120.

Most central site ISDN equipment can operate in either V.120 or PPP. While PPP offers somewhat better throughput, because of its lower overhead, the difference is very small. Encapsulating PPP within V.120 can be done but does not provide the most efficient use of bandwidth.

H channels

One advantage for PRI users is the ability to place a single call and bundle multiple B channels together. When B channels are bundled for a call it is referred to as an H channel. For PRI there are 4 standard H channels. As you will see by the descriptions some H channels can only work with E1 rates and others only make sense for Tl rates.

HO channels operate at 384 KBPS. This speed is achieved by bundling 6 B channels for one call. The history behind this is related to Video Conferencing. 384Kbps is the most cost effective method of doing video conferencing. To substantially improve performance it would take a significant increase in the number of B channels at additional cost. Below this speed quality drops off sharply. HO makes sense for both E1 and Tl based PRI.

H11 channels operate at 1.536 MBPS. If all 24 B channels on an interface are combined you get this speed. One problem is that there is no time slot available for the D channel. The Idea is that the D channel will reside on another physical interface. This requires the user to purchase two PRI interfaces with one D channel controlling all of the B channels on both physical interfaces.

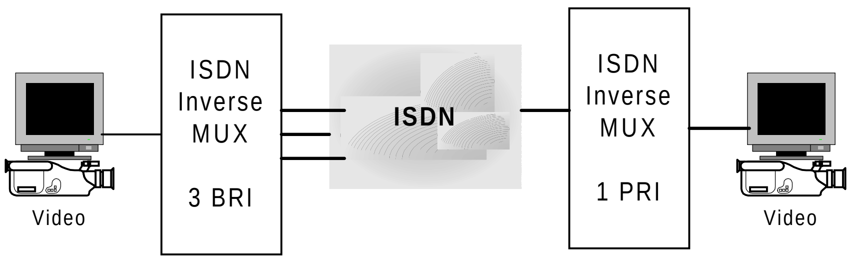
H12 channels operate at 1.920 MBPS, which are all 30 channels of an E1 PRI. Unlike H11the D channel resides on the same physical interface.

H10 channels operate at 1.472 Mbps. To do this all 23 B channels on one PRI interface are combined. While I have no proof I would guess that the H10 came about as a result of users unwilling to pay for 2 PRIs just to get an H11 channel. Many documents don't discuss it at all which leads me to believe it came about after the other H channels were agreed upon.

H channel support from carriers varies widely. I am unaware of Net 3 countries supporting H10 or H11 channels. That doesn't mean that they don't, I just haven't seen it. The only universal H channel is HO at 384 KBPS (6 B channels).

While in T1 rate PRI HO, H10 and H11 channels are defined for North American they are not available from all carriers. AT&T has made them available in the US but other PRI providers may not support them. In the event H channels are not supported on your circuits there are other options such as inverse multiplexing or bonding that can be used to achieve high bandwidth connections.

Inverse Multiplexing performs a similar function to H channels, that is, combining several channels into one data stream. The significant difference is that it is the user equipment on each end of the circuit that performs the function. From the phone company perspective a series of phone calls have been placed between two locations, the fact that they are related to one data stream is unknown to the carrier. Because this requires no service provider intervention it is the most common method of video conferencing.



ISDN Video conferencing

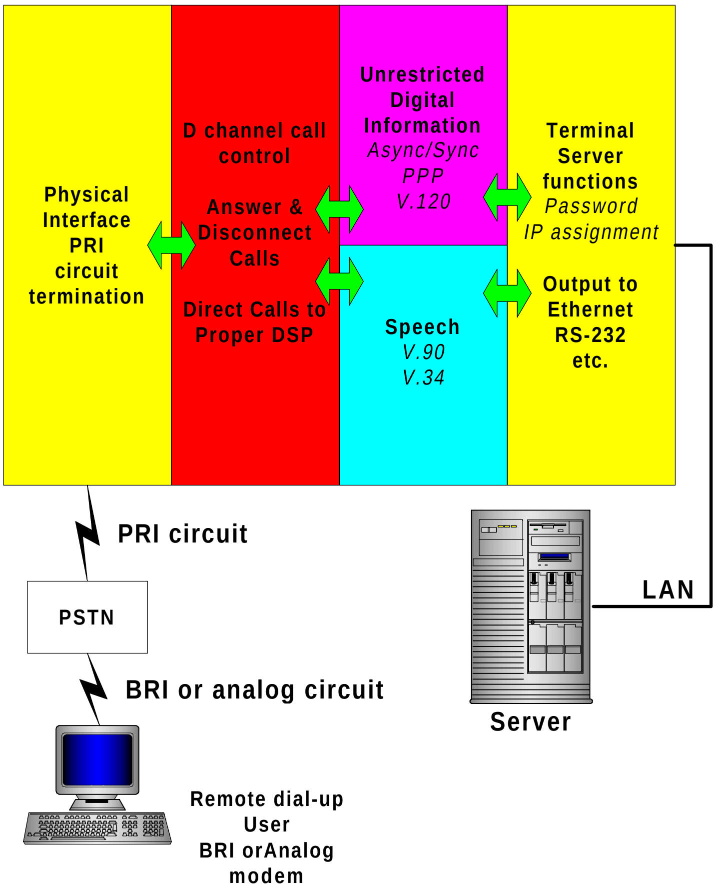
Figure 34

Figure 34 shows an example of inverse multiplexing for video conferencing. The remote location on the left has a video codec and an inverse mux capable of having three BRI circuits attached. To setup a videoconference with the site on the right it will place 6 data calls, one call on each B channel. The answering site contains a compatible piece of multiplexing equipment with a PRI interface. The six calls will come in and the user equipment will perform a training routine. Once the training is complete each side is able to split data stream between the six connections. On the receive side it is able to combine the six data streams into one. The net result, assuming 64Kbps B channels, is a 384Kbps full duplex connection.

This method is probably the most commonly employed way to engage in videoconferencing. One of the drawbacks of this method when compared to the H channel is that the user equipment is more complex. In the network shown above the user equipment must make 6 separate calls and perform the alignment procedure. The advantage is that no special telephone network support is required. It would therefore be possible to replace the PRI interface with a standard T1 interface, or likewise, to replace the BRI site with six Switched 56 (DDS based) circuits. As long as the two sites can call each other many variants in switched network access can be employed.

PRI remote access

PRI is a commonly used interface for remote access. Properly designed access equipment allows great flexibility in dealing with both ISDN and analog modem calls. Shown in figure 35 is the Basic concept behind how a PRI interface deals with an incoming call. The D channel subsystem will do all the communication with the service provider.



PRI terminal server functions
Figure 35

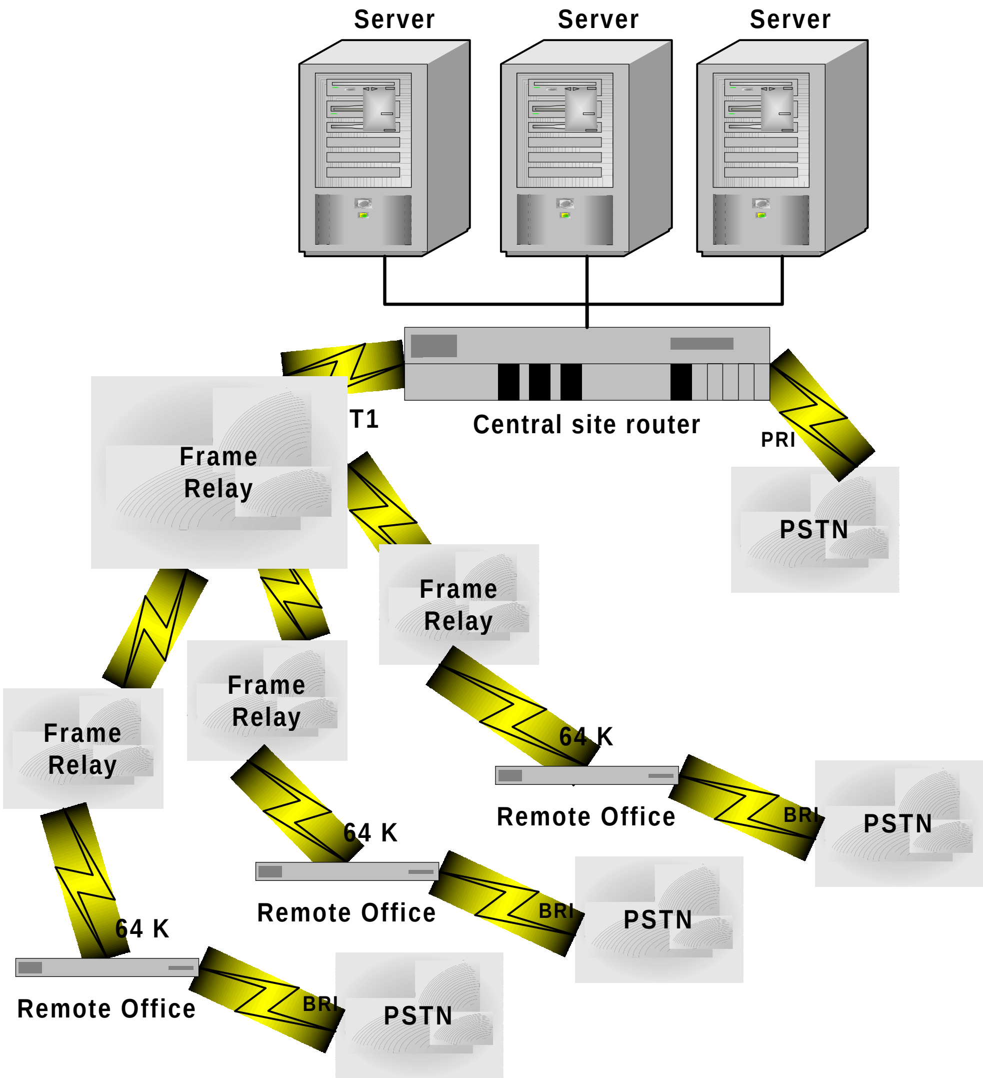
When an incoming setup message is seen on the D channel it will be examined to determine its validity. The determination of a valid call is based upon the capabilities and configuration of the user’s PRI equipment. If the call is accepted the controller must direct the B channel data stream to the appropriate device to properly interact with the calling equipment. As an example data calls from ISDN Terminal Adapters may be PPP encapsulated IP. In this case the PRI equipment must be able to negotiate PPP operating parameters and function in a pure digital environment.

If the incoming call is a speech call from a modem it will require the PRI device to engage in PCM representation of the analog training sequence, effectively imitating a modem. The analog signal will exist only on the local loop serving the remote user. DSP technology within the PRI device allows a pure digital representation of all modem signals. This is the only way V.90 modems can operate at higher speeds.

Frame Relay backup

One of the most common applications for ISDN is in backup to frame relay router circuits. Typically centrally located routers have one or more frame relay interfaces. Each DLCI represents a connection to a router at a remote location.

Typically the remote locations have lower speed interfaces than the central site. Designing a dial backup system for this type of network can be effectively done using a combination of PRI at the central location and BRI at the remote locations. Figure 36 below shows how this type of installation can be done.



Router backup using BRI and PRI
Figure 36