DEFENSE COMMUNICATIONS ENGINEERING CENTER

STU-III SYSTEM LEVEL DESCRIPTION

AND

NETWORK APPLICATIONS

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This paper serves as the system level description of the Secure Terminal Unit Type III (STU-III) and its interfaces in a variety of DoD switched voice network applications. The STU-III terminal is being developed by the National Security Agency ~~(NSA)~~ and is intended to replace several thousand existing analog clear voice telephone instruments in the DoD. Its operation and interface to the switched networks is essentially the same as for existing or planned clear voice telephones. The STU-III family of terminals includes two types of Low Cost Terminals (LCT-1 and LCT-2), a STU-II compatible terminal (STU-III/A) and a ruggedized mobile/portable terminal (STU-III/MPT). The LCT-1 is intended to be the primary DoD secure voice terminal, whereas the STU-III/A and STU-III/MPT will be used for special applications, e.g., for interoperability with the NATO STU-II terminals and tactical operations requirements.

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STU-III SYSTEM LEVEL DESCRIPTION AND NETWORK APPLICATIONS

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SECTION 1 - INTRODUCTION

This paper serves as the system level description of the Secure Terminal Unit Type III (STU-III) and its interfaces in a variety of DoD switched voice network applications. The STU-III terminal is being developed by the National Security Agency (NSA) and is intended to replace several thousand existing analog clear voice telephone instruments in the DoD. Its operation and interface to the switched networks is essentially the same as for existing or planned clear voice telephones. Installation of a STU-III in the United States can be very simple; normally, installation for a two-wire application consists only of inserting a telephone connector (e.g., RJ11C) and an AC power plug into the appropriate receptacles. The STU-III family of terminals includes two types of Low Cost Terminals (LCT-1 and LCT-2), a STU-II compatible terminal (STU-III/A), and a ruggedized mobile/portable terminal (STU-III/MPT). The STU-III/LCT-1, STU-III/A, and STU-III/MPT are intended for all levels of government classified traffic. The STU-III/LCT-2 is intended for sensitive unclassified traffic. The LCT-1 and the LCT-2 units will be available in at least three vendor variations; i.e., AT&T, Motorola, and RCA. The STU-III/A is in the Low Rate Initial Production (LRIP) phase, deliveries are expected by October, 1989 from Motorola. The MPT is currently in the development phase with Motorola. A production contract is anticipated in early 1989. All STU-III terminals are interoperable at 2.4 kb/s using an enhanced Linear Predictive Coder voice algorithm. The LCT-1 is intended to be the primary DoD secure voice terminal, whereas the STU-III/A and STU-III/MPT will be used for special applications (e.g., for interoperability with the NATO STU-II terminals and tactical operations requirements.

SECTION 2 - STU-III PHYSICAL DESCRIPTION AND BASIC FEATURES

2.1 PHYSICAL DESCRIPTION

The STU-III consists of a deskset, handset, and, depending upon the vendor and/or location, a separate AC power adapter. The approximate range of physical parameters is: 225-900 cubic inches, 8-15 lbs., and 14-30 watts. This range of parameters varies respectively in size, weight, and power depending upon the vendor and the type of terminal provided (i.e., STU-III/LCT, STU-III/A, or STU-III/MPT).

2.2 STU-III FEATURES AND OPTIONS

The STU-III terminals will incorporate the following features and options:

1. STU-III/LCT-1

   a. Common Features

      (1) Clear and secure voice operating modes

      (2) Digital voice processing based on Linear Predictive Coding, Enhanced (LPC-10e) at 2.4 kb/s for the secure mode

      (3) Full-duplex operation over two-wire access lines, with internal echo cancellation

      (4) Half-duplex operation (VOX) over two-wire access lines

      (5) Secure data capability at 2.4 kb/s, synchronous.

      (6) EMI/TEMPEST protection

      (7) Crypto keying using the FIREFLY II algorithm

      (8) Precedence tone dialing for two- and four-wire applications

(9) Preempt tone recognition

(10) Secure dialing mode, which allows address information to be passed securely to a distant interface, e.g., conference bridge or RED switch.

b. Common Options

(1) Foreign country operation by meeting electrical, functional, and power requirements of host country

(2) Four-wire AUTOVON/DSN operation

(3) Abbreviated dialing and storage of several frequently called telephone numbers

(4) HEMP protection

(5) Multiline compatibility with industry standard (Bell System's 1A2) Key Telephone System.

c. Vendor Available Options

(1) Dual Homing capability implemented via four-wire AUTOVON/DSN and two-wire DSN/public switched networks (RCA)

(2) Digital voice processing at 4.8 kb/s using non-LPC algorithm (AT&T)

(3) Secure and nonsecure data options (See Table 1)

(4) Cellular radio model including cellular radio with capabilities for signaling, vehicular mounting, and power (Motorola)

(5) RED Interface Terminal (RIT) capabilities including the capability to remote the RED analog and RED digital voice outputs, deskset dialing, and control functions to another location [The basic AT&T and RCA terminals

Table 1. Vendor Data Port Modes (LCT-1)

SPEED	MODE	FEC[1]	AT&T	MOT	RCA
4800	SYNC	YES			
		NO	X[2]		
	ASYNC	YES			
		NO	X[2]		
2400	SYNC	YES			
		NO	X[2]	X	X
	ASYNC	YES			
		NO	X[2]	X	X
1200	SYNC	YES			
		NO			
	ASYNC	YES	X[2]		X
		NO		X	X
600	SYNC	YES			
		NO			
	ASYNC	YES			X
		NO			X
300	SYNC	YES			
		NO			
	ASYNC	YES			X
		NO		X	X
150	SYNC	YES			
		NO			
	ASYNC	YES			X
		NO			X
75	SYNC	YES			
		NO			
	ASYNC	YES			X
		NO			X

NOTES: 1. FORWARD ERROR CORRECTION
2. CLEAR DATA CAPABILITY

have this capability; Motorola has a separately configured, rack-mountable terminal, Automatic Remote Secure Telephone Unit (ARSTU), to provide this capability. The remote interface for each vendor's RIT is different and is defined in separate interface control documents.]

(6) Plain Old Telephone Service (POTS) during local AC power failure (Motorola)

2. STU-III/A - The STU-III/A has the same common features and options, except for the multiline option, of the STU-III/LCT-1, plus the following:

   a. Secure Asynchronous Data at 300, 1200 and 2400 b/s

   b. STU-II BELLFIELD keying capability

   c. STU-II NET mode

   d. STU-II NET MULTIPOINT mode. [The NET MULTIPOINT (NET BROADCAST) mode of operation provides for half-duplex push-to-talk, broadcast communications between two or more STU-III/A's holding common NET keys.]

3. STU-III/MPT - The STU-III/MPT has the same common features and options, except for the multiline option, of the STU-III/LCT-1 plus the following:

   a. Secure Asynchronous Data at 300, 1200 and 2400 b/s

   b. STU-II NET MULTIPOINT mode

   c. Black digital interface (RS-449/232/MIL-STD-188C) to connect to an external device such as a modem

   d. Cellular interface directly compatible with the MOTOROLA 6000 Mini-Tac cellular transceiver

   e. Interface to external military radios

f. STU-III Dedicated mode with ring and answer capability for "hot line" applications [Either a switch connection must be held or a dedicated circuit must be provided in order to effect this mode of operation. This mode, once established, will allow the users to physically go on-hook while still maintaining the connection. When either user goes off hook and selects clear or secure, a ring indication is sent to the distant terminal alerting the user of an incoming call.]

SECTION 3 - SYSTEM OVERVIEW, INTERFACES, AND CALL PROCEDURES

3.1 SYSTEM OVERVIEW

Figure 1 shows the various applications of the STU-III in the Defense Communications System (DCS) and other networks, either as an end terminal on the backbone network switches and PBX's, or as a RED Interface Terminal (RIT) for interfaces to other networks. As shown, the STU-III in its RED Interface Terminal configuration will be integrated into the RED switch and the STU-III conference bridge to provide access for these devices to the switched voice networks and to other STU-III's or interface devices. The interconnection matrix for the STU-III's is shown in Table 2. As indicated, all STU-III's can directly interoperate with each other via the switched networks. Only the STU-III/A can directly connect via the DSN/IVSN networks to a NATO KY-71 (STU-II). A manual Radio Wireline Interface (RWI) will be required to allow the STU-III to interoperate with tactical VINSON/ANDVT terminals. Direct automatic interoperability will be possible with the RED switch and the STU-III conference bridges via the RIT. Further explanation of these interconnections and associated call procedures is provided in subsequent paragraphs. It is intended that the STU-III will utilize existing voice grade telephone access lines; new facilities, if required, need not differ from those required by a normal telephone installation.

3.2 STU-III NETWORK INTERFACES

As indicated, the STU-III (except for the MPT) is a replacement for existing clear voice telephone terminals. In general, the STU-III's have two basic termination interfaces - i.e., two- and four-wire - applicable to the switched voice DoD, U.S., and foreign public networks. These are shown in Figures 2 and 3. The two-wire mode is full-duplex with a manual fallback to half-duplex. The predominant utilization of the STU-III will be

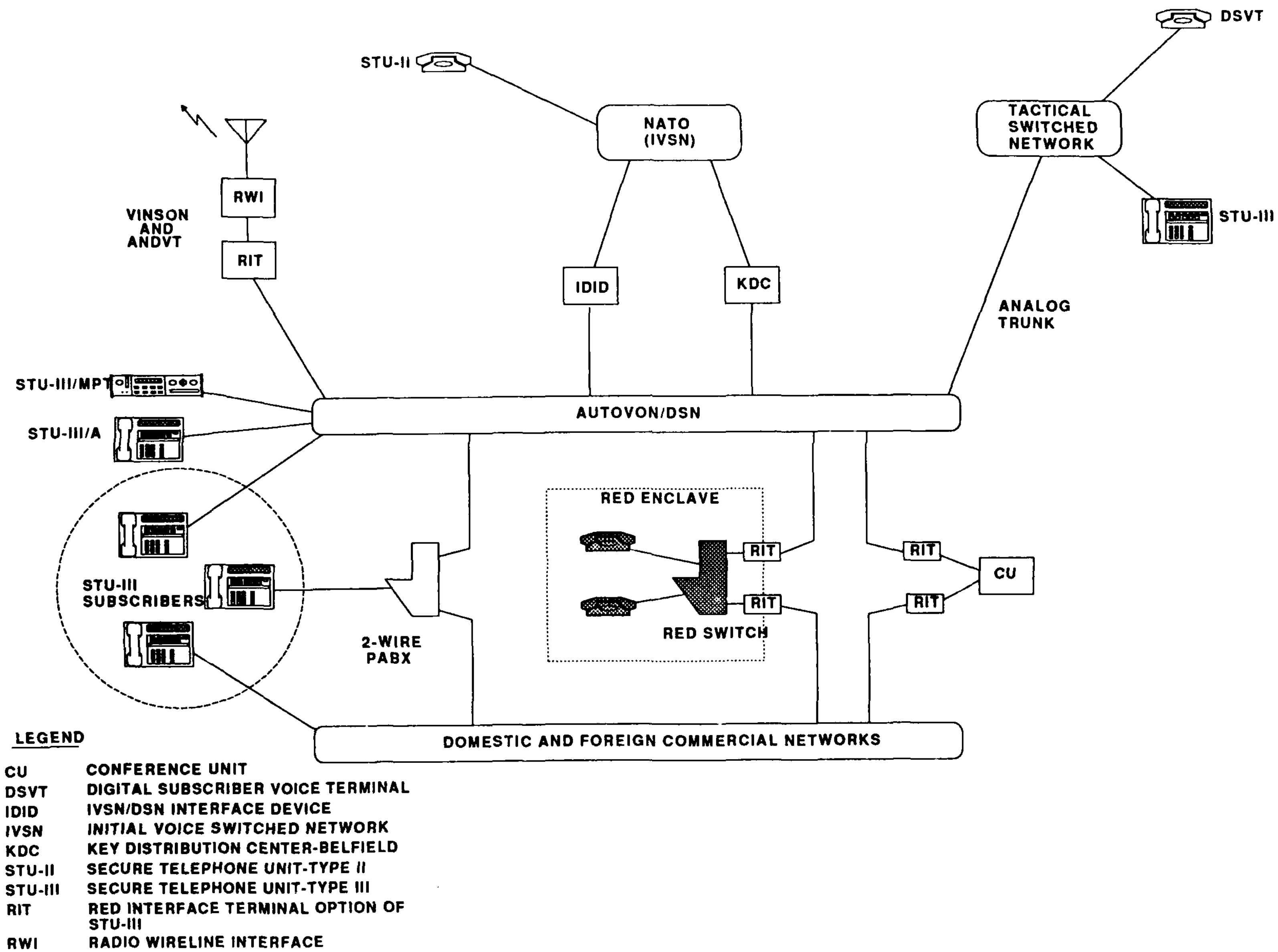


Figure 1. STU-III Network Applications

Table 2. STU-III/LCT-1, STU-III/A and STU-III/MPT Interconnection Matrix

	STU-III/LCT-1	STU-III/A	STU-III/MPT	RED SWITCH	CONFERENCE UNIT	VINSON (AIR/SHIP)	ANDVT (AIR/SHIP)	STU-II (NATO SY-71 OR SPENDEX-40)	TRI-TAC
STU-III/LCT-1	DIRECT	DIRECT	DIRECT	AUTOMATIC (RA/RD)	MANUAL (RD)	RWI (RA)	RWI (RD)	NONE	(Note 3)
STU-III/A	DIRECT	DIRECT	DIRECT	AUTOMATIC (RA/RD)	MANUAL (RD)	RWI (RA)	RWI (RD)	DIRECT (Note 2)	(Note 3)
STU-III/MPT	DIRECT	DIRECT	DIRECT	AUTOMATIC (RA/RD)	MANUAL (RD)	RWI (RA)	RWI (RD)	DIRECT (Note 1)	(Note 3)

LEGEND

RA - RED ANALOG BREAKOUT FROM 2.4 kb/s LPC
RD - RED DIGITAL 2.4 kb/s LPC
RWI - RADIO WIRELINE INTERFACE (MANUAL)

NOTE

1. CAN INTEROPERATE IN STU-II MULTIPOINT COMSEC MODE ONLY.
2. CAN INTEROPERATE IN ONE OF THREE STU-II COMSEC MODES; KDC, NET AND MULTIPOINT. KDC MODE REQUIRES ACCESS TO BELLFIELD KDC FOR KEY I.E. NATO STU-II's (SEE FIG 9).
3. VIA TACTICAL DEPLOYMENT OF STU-IIIs OR RED SWITCH (SEE PARA. 3.7).

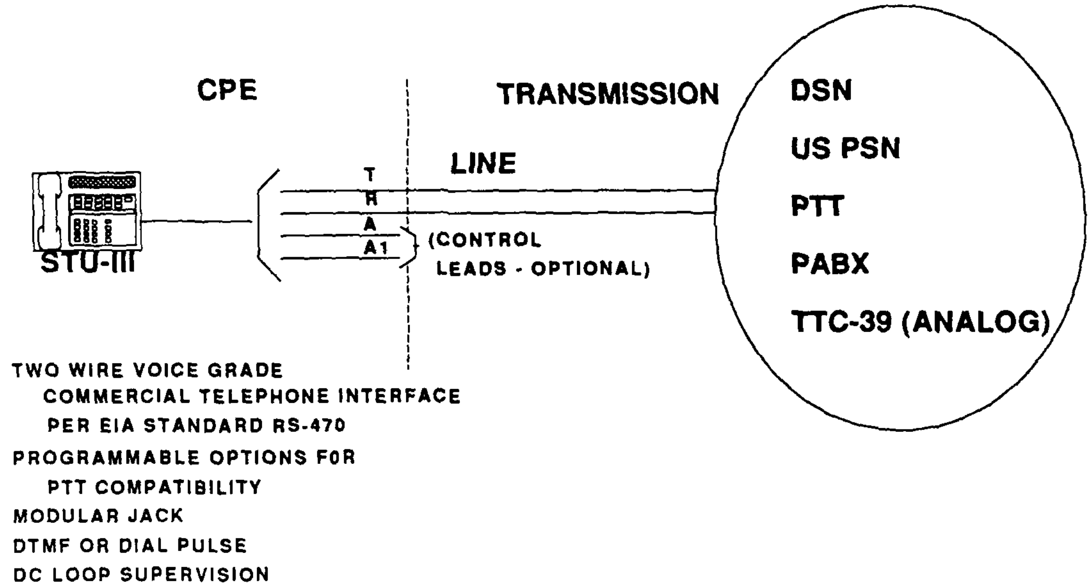


FIGURE 2A. A SINGLE LINE INTERFACE

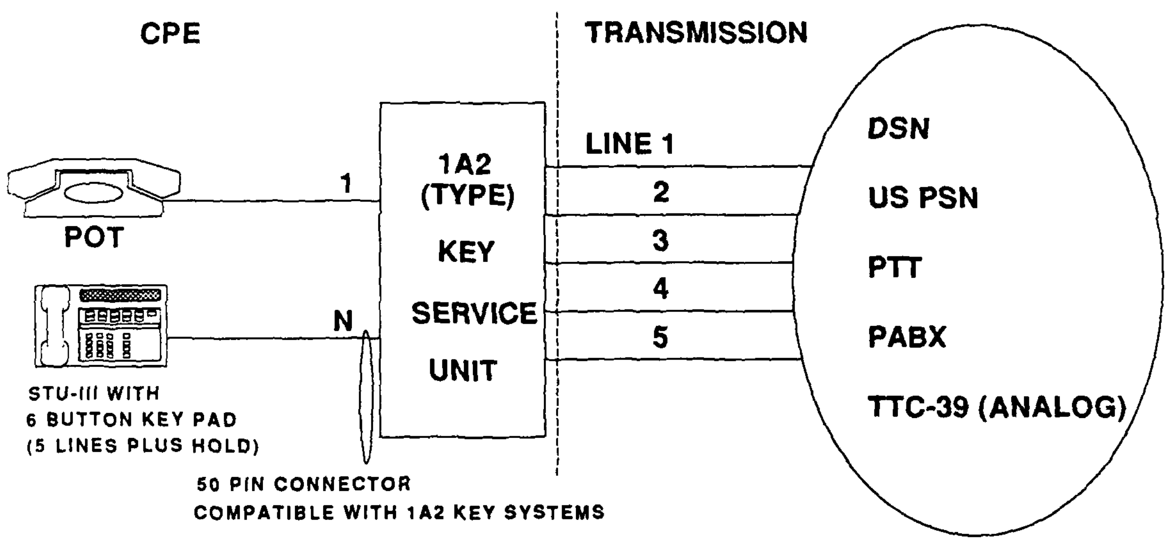


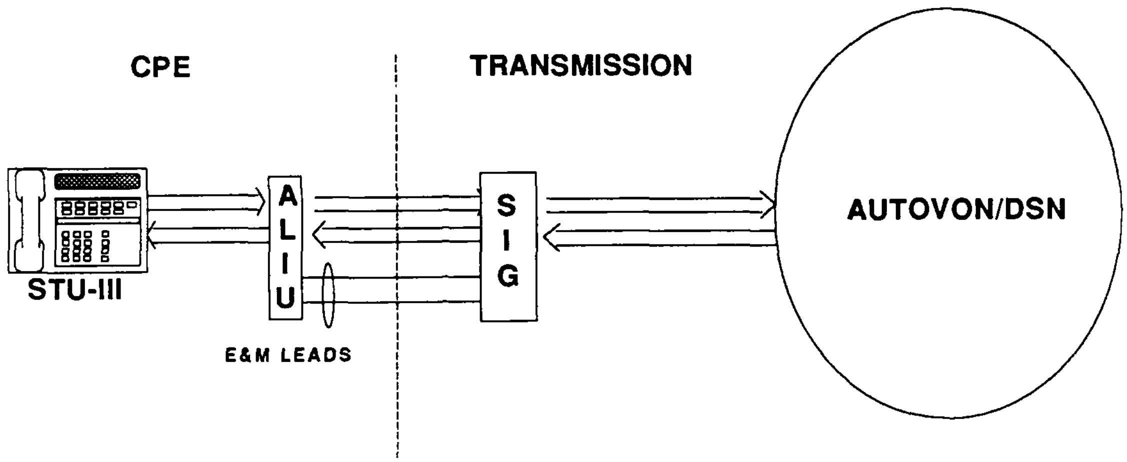
FIGURE 2B. A MULTI-LINE INTERFACE

LEGEND

CPE CUSTOMER PREMISE EQUIPMENT

POT PLAIN OLD TELEPHONE

Figure 2. STU-III Two-Wire Network Interface



4-WIRE SUBSCRIBER LINE
INTERFACE PER DCAC 370-V175-6/13
DTMF DIALING
E & M SUPERVISION

LEGEND

ALIU AUTOVON LINE INTERFACE UNIT
CPE CUSTOMER PREMISE EQUIPMENT
SIG SIGNALING UNIT

Figure 3. STU-III Four-Wire Network Interface

in the two-wire full-duplex mode. The interfaces shown are applicable to all vendor models of the STU-III. The two-wire interface shown in Figure 2 is the standard commercial interface for single line two-wire telephone lines in accordance with EIA Standard RS-470. For overseas applications, the terminal interface will be programmable and adaptable to meet foreign public switched networks (PSN) variations, for example, signal delays or levels. This interface will allow connection to the AUTOVON, DSN, PSN's, PABX's and the AN/TTC-39 analog matrix. The STU-III's four-wire interface shown in Figure 3 is identical to the standard AUTOVON/DSN interfaces as specified in DCAC 370-V175-13. Currently, the AUTOVON four-wire terminals (with Multilevel Precedence and Preemption (MLPP) capability) are primarily for C^2 subscribers; some will be replaced in the initial application of STU-III's. In general, the four-wire STU-III will require a STU-II AUTOVON Line Interface Unit (ALIU). However, each installation must be considered on a case-by-case basis. The NSA STU-III TELCO Connection and General Information Report, 4 March 1988, provides specific details on each type of ALIU that will work with the STU-III and how each is to be wired. This document also provides interface details on how to connect various terminal interfaces (i.e., two-wire, four-wire, and multiline key systems). Tables 3 and 4 show the various STU-III/LCT wireline interfaces and interface configurations.

3.2.1 Secure Nonsecure Call Features

Depending on the type central office, various features are provided to the user by modern automatic central office equipment that supports call processing. These features will vary based on the type of central office or PBX to which the STU-III is connected. However, there is a basic set that is generally found on modern equipment and is specified in the DSN Generic Switch Specification. Of these features, all will work with the STU-III in the clear mode. In the secure mode, conferencing (to include

Table 3. STU-III LCT-1 Application Configurations

CODE	CONFIGURATION	USAGE
C2W	CONUS 2-wire	Standard phone and 2-wire AUTOVON
C4W	CONUS 4-wire AUTOVON with MLPP and HEMP	CONUS MLPP AUTOVON circuits
CDH	CONUS dual-homed	Connect to standard phone and 4-wire MLPP AUTOVON at same time
CMK	CONUS Multiline Key System (*IA1/IA2*)	Up to 5 standard 2-wire lines
F2W	Foreign single line 2-wire	Standard foreign phone systems
F4W	Foreign 4-wire AUTOVON with MLPP and HEMP	Foreign MLPP AUTOVON circuits
FDH	Foreign dual-homed	Connect to standard foreign systems and 4-wire MLPP AUTOVON at same time
FMK	Foreign Multiline Key System	Up to 5 standard foreign 2-wire lines

LEGEND
MLPP - Multilevel Precedence Preemption
HEMP - High Altitude Electromagnetic Pulse

Table 4. LCT-1 Interface Configurations

TERMINAL CONFIGURATION	CIRCUIT TYPE	TERMINAL INTERFACE	USER PROVIDED INTERFACE	POWER (VAC)	COMMENT
C2W	CONUS Commercial	RJ11C	625T Compatible	110	
F2W	Foreign Commercial	RJ11C	RJ11C Compatible	220	
C2W or F2W	Analog PBX	RJ11C	RJ11C Compatible	110/220	
C2W or F2W	Digital PBX	RJ11C	RJ11C Compatible	110/220	Change digital subscriber interface card to an analog single line interface card.
CMK or FMK	Multiline	1A1/1A2 (50PIN)	RJ24X Compatible (50PIN)	110/220	Implement automatic exclude and restore or manual exclude and restore function of the local key set.
CMK or FMK	Multiline using single-line instrument	RJ13C	RJ13C	110/220	RJ13C interface must be provided for all single line instruments that operate on a multiline key set. When single line instruments are installed on a 1A1/1A2 key system, the single line instrument must be capable of A lead control.

Table 4. LCT-1 Interface Configurations (Cont'd)

TERMINAL CONFIGURATION	CIRCUIT TYPE	TERMINAL INTERFACE	USER PROVIDED INTERFACE	POWER (VAC)	COMMENT
C2W or F2W	AUTOVON	2-wire RJ11C	2-wire	110/220	Regular PBX with a dial"8" capability.
C4W or F4W	AUTOVON	4-wire	4-wire	110/220	4-wire AUTOVON access: programmable AUTOVON exchange (334 or compatible) with a precedence and preemption capability is required.
C4W or F4W	AUTOVON	4-wire	6/8-wire e.g. RJ2GX	110/220	6/8-wire AUTOVON access: AUTOVON Line Interface Unit (ALIU) is required.
CDH or FDH	Dual Homed	RJ11C plus 4-wire AUTOVON		110/220	Call interrupt feature is required.

three party), call waiting, precedence call waiting, and attendant busy override features will not work. All secure conference calls must use a STU-III digital conference bridge (see paragraph 6). The other features listed will not work because their associated tones are not audible to the STU-III user when the terminal is in the secure mode. A preempt tone will be detected and both visually and audibly indicated to the user by the STU-III.

3.3 STU-III to STU-III CALL PROCEDURES

Dialing procedures for the STU-III are similar to those used for the normal telephone. Calls are established as plain analog connections through the network through use of the DSN numbering plan and call procedures. The user goes off-hook, dials precedence (if applicable), dials the called party number, converses with the called party in the clear mode, and then selects the secure mode. Synchronization for the secure mode will then be accomplished by the STU-III's and will take about 15 seconds. Normally, all calls are automatically established as full-duplex; however, half-duplex operation may be selected manually as a backup when telephone lines are encountered that do not support full-duplex operation, such as excessive echo or echo suppressors that cannot be disabled.

3.4 INTERFACE TO RED SWITCHES

RED switches provide secure voice service through use of physically secure switching and distribution facilities. Both existing and new DoD RED Switch Project (RSP) switches that require access to the DSN will be equipped with STU-III secured interfaces. These interfaces may operate in either a trunk mode or a user access line mode to DSN.

3.4.1 Trunk Interface Mode

This interface mode provides automatic network in and out dialing between the RED Switch and DSN, and is the preferred method of operation for the RSP switches.

3.4.1.1 Interface Description

The RED Switch will appear as an End Office (EO) interface to the DSN (see Figure 4). Each trunk will be equipped with a RED Interface Terminal (RIT) and an interface applique function to be provided integral to the RED Switch. The RIT is a configuration of the STU-III that provides remotable RED digital audio, control, and status inputs/outputs. The applique function provides compatible interfaces to the RIT, the RED Switch trunk port, and the DSN trunk. The applique also provides monitoring and control of the RIT; provides RED/BLACK isolation; and provides processing of signaling, supervision, and addressing information between the RED Switch and BLACK DSN trunk.

3.4.2 <u>RED Switch Interface Call Descriptions for Trunk Interface Mode</u>

3.4.2.1 RED Switch User to STU-III

The calling RED Switch user dials a precedence digit, a two-digit access code, and the seven-digit DSN number of the called STU-III. The switch seizes an idle interface trunk and forwards the call data to the interface applique which isolates and forwards the call data to DSN. The DSN then completes the connection to the distant STU-III. When the called STU-III answers, both the RIT and the called STU-III go to the secure mode. The RED Switch then connects the calling RED Switch user to the interface trunk, allowing secure communications to begin.

3.4.2.2 STU-III to RED Switch User

The calling STU-III dials a precedence digit and the seven-digit DSN number of the called RED Switch user. The DSN routes the call to an idle RED Switch trunk and forwards the call precedence and addressing information to the interface applique. The interface applique in turn isolates and forwards the information to the RED Switch for processing. The RIT and calling

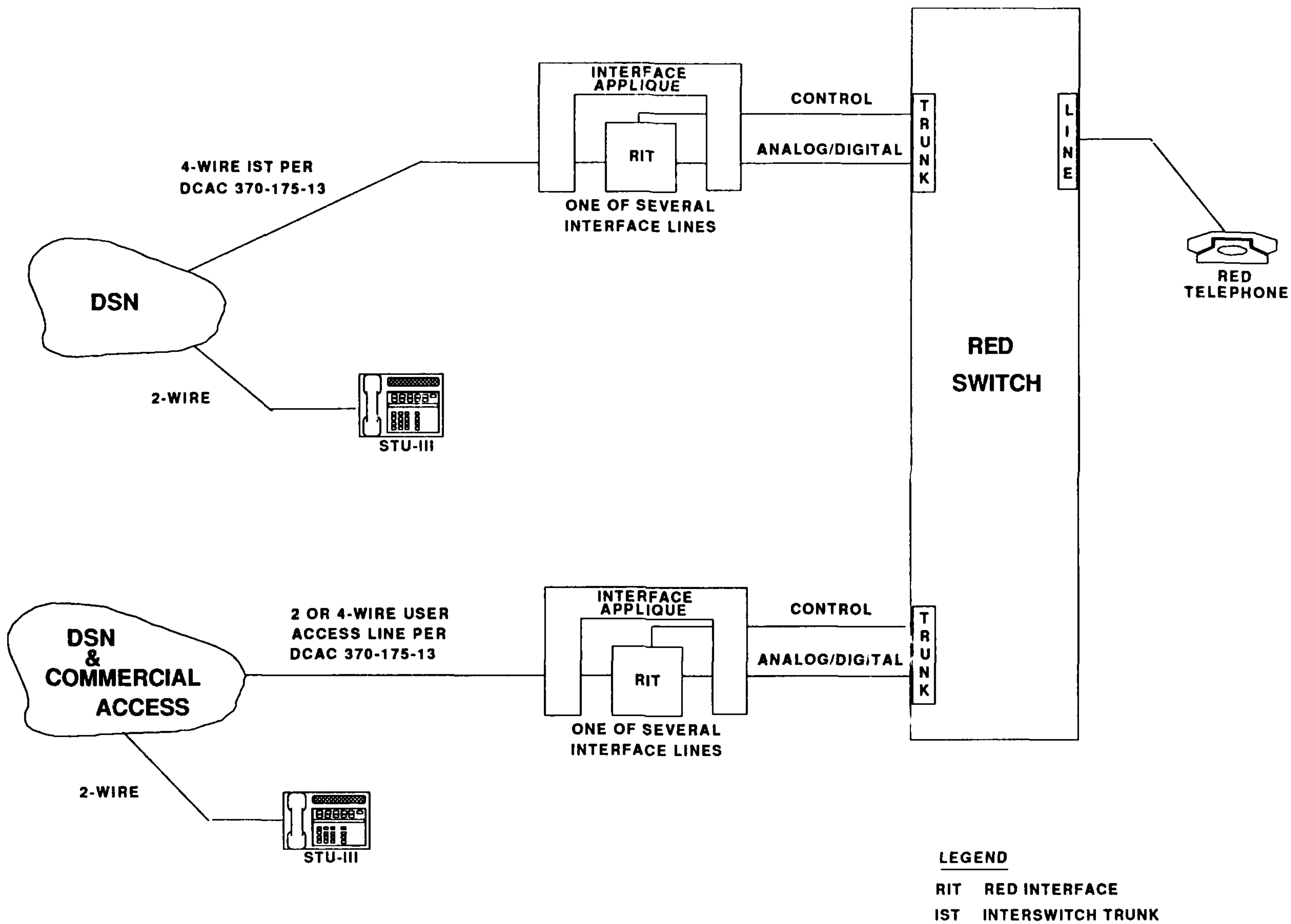


Figure 4. RED Switch Interface to Switched Networks

STU-III go to the secure mode automatically, and the RED Switch rings the called party. The RED Switch completes the connection when the called party answers; secure communications may then begin.

3.4.3 User Access Line Interface Mode

The capability will also be provided at the RED Switches to provide a user access line interface to the DSN from the RED Switch, either two- or four-wire, via the RIT and associated interface applique. In this case the RED Switch looks like a user station to DSN, as opposed to another switch. This interface could be used in cases where automatic network in-dialing with precedence is not required, e.g., access through the base PABX.

Outgoing calls may be either operator-assisted or automatic from the RED Switch user. Incoming calls may be operator-assisted in the secure mode or automatic when the calling STU-III is in the Secure Dial mode. Any or all of these options are available in the RSP switches.

3.5 INTERFACE TO DSN STU-III CONFERENCE UNITS

STU-III users will be provided a secure conferencing capability as a service feature of the DSN. Initially, the capability will be provided by three geographically distributed, operator-controlled AN/FTC-52 digital conference units. The conference units will employ a RED digital broadcast conference protocol. This digital technique eliminates the need for RED analog tandem connection of LPC-processed speech between conferees and the associated voice quality degradation which would occur. Conference units will be interfaced to the DSN through RED Interface Terminals (RIT's). One conference unit is planned to be provided in the CONUS, one in Europe, and one in the Pacific to provide an operational capability and to gather usage data to further define general purpose STU-III conferencing requirements for an expanded STU-III population.

3.5.1 Interface Description

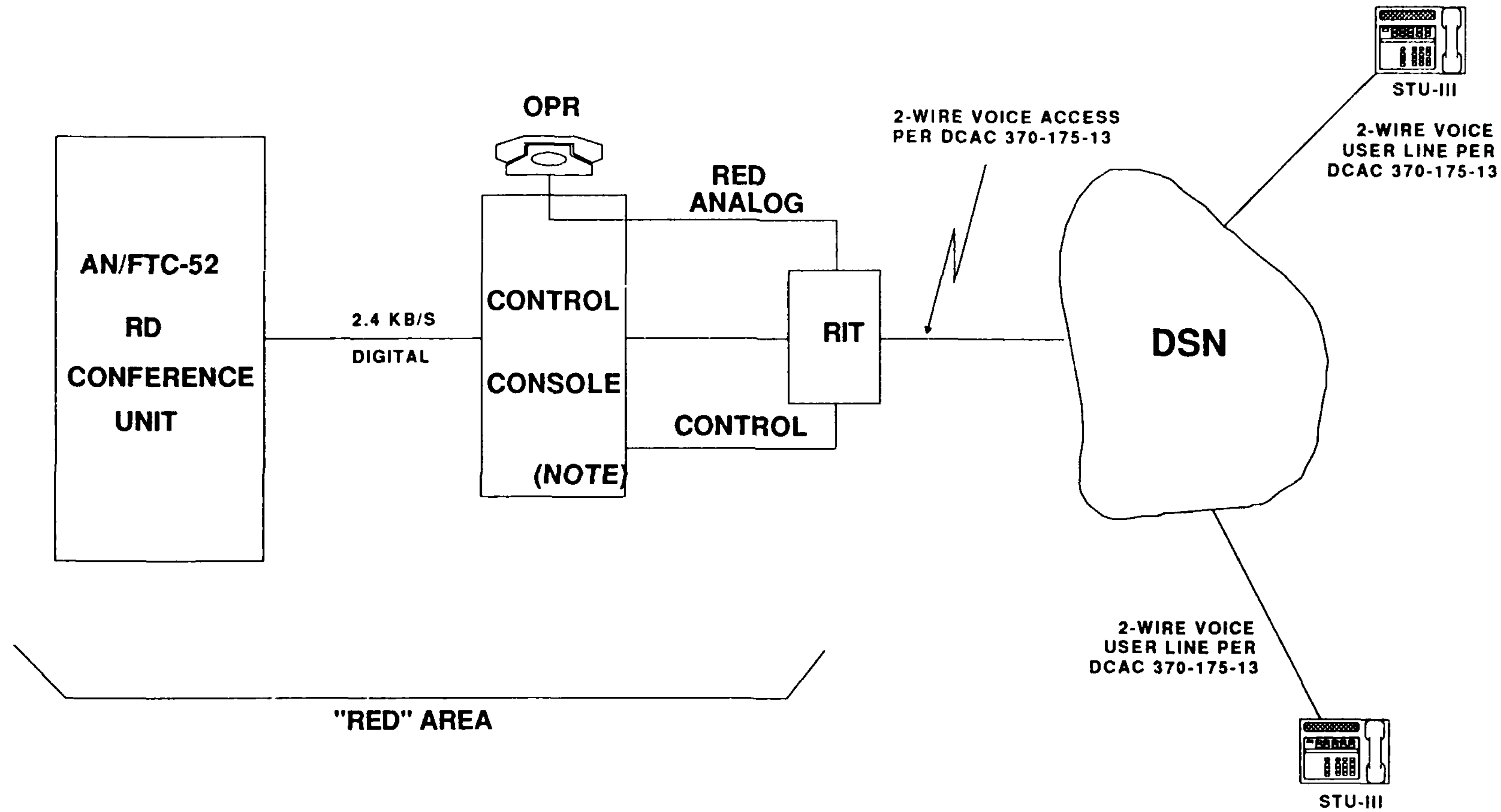
The conference unit interface to the DSN (see Figure 5) will consist of a number of two-wire DSN voice user lines. Each line will be equipped with an RIT interfaced to an operator's conference control console. The control console will provide the operator with the capability to control and monitor the RIT and AN/FTC-52; provide RED/BLACK isolation; and process signaling, supervision, and addressing between the conference unit and BLACK DSN line. The lines will be placed in a common hunt group within DSN such that only one telephone number is required to access the conference unit.

3.5.2 Conference Call Description

To initiate a conference, the conference originator dials a precedence digit and the seven-digit DSN number of the conference unit location. DSN completes the connection to the conference unit, the call is answered by the conference unit operator, and both the RIT and the calling STU-III go to the secure mode. The originator then provides the operator with the telephone numbers of the desired conferees. The conference unit operator then calls each conferee through separate RIT's. When each conferee answers, the operator announces the conference, then both the RIT and the called STU-III go to the secure mode and the operator connects the conferee to the conference unit. Once all conferees have answered, the conference may begin. During the conference, the conference unit monitors the incoming RED digital signal from each interface line to detect speech activity. The active (first active if more than one) speaker's digital input is then broadcast to all other conferees. Once the active speaker yields, the conference unit is available to be seized by a responding conferee.

3.6 TACTICAL RADIO/WIRELINE INTERFACES (RWI's) TO ANDVT/VINSON

UHF, VHF, and HF radio and satellite broadcast networks provide connectivity between airborne and naval forces, and access



NOTE

THE CONTROL CONSOLE AND AN/FTC-52
CAN HANDLE 10 RITs/ACCESS LINES

LEGEND

RIT RED INTERFACE TERMINAL
RD RED DIGITAL

Figure 5. Current General Purpose STU-III Conference Approach

to the DCS switched networks via operator-assisted radio/wireline interface stations. Network secure voice services will be provided by the 16 kb/s VINSON and the 2.4 kb/s ANDVT, which are being introduced into these networks. The STU-III terminal will employ a common analog-to-digital algorithm at 2.4 kb/s with the ANDVT; thus, they will provide a good speech quality RED digital connection to the ANDVT. Connections to VINSON terminals will require an analog breakout between the 2.4 kb/s STU-III terminal and the 16 kb/s VINSON. Radio/wireline interface stations may be equipped with one or more STU-III RED Interface Terminals (RIT's) to provide secure access to and from STU-III users (see Figure 6). Interface consoles will be developed by the cognizant military departments to provide an operator-controlled interface between the RIT's and the secure radio equipment at each station. The capabilities required of the interface console will vary between RWI stations based on the number of access lines required and the number and type of secure radio equipments provided. The description below covers the generalized case with multiple access lines, each capable of being connected either to an ANDVT via a RED digital interconnect or to any secure link via a RED analog interconnect.

3.6.1 Generalized RWI Description

The RWI will consist of RIT's for STU-III interoperability, an operator console for establishment and control of connections, and network radio COMSEC as required for interoperability with the radio networks being served. Optionally, each RIT may connect to the switched networks via a four-wire AUTOVON/DSN line or two-wire facilities. The interface console will provide two basic types of connections: (1) RED digital connections between STU-III's and ANDVT's; and (2) RED analog connections between STU-III's and VINSON's. The console will interface the RED KG, digital speech, analog speech, control, and status input/outputs of each RIT and will contain the necessary circuitry to monitor and control each

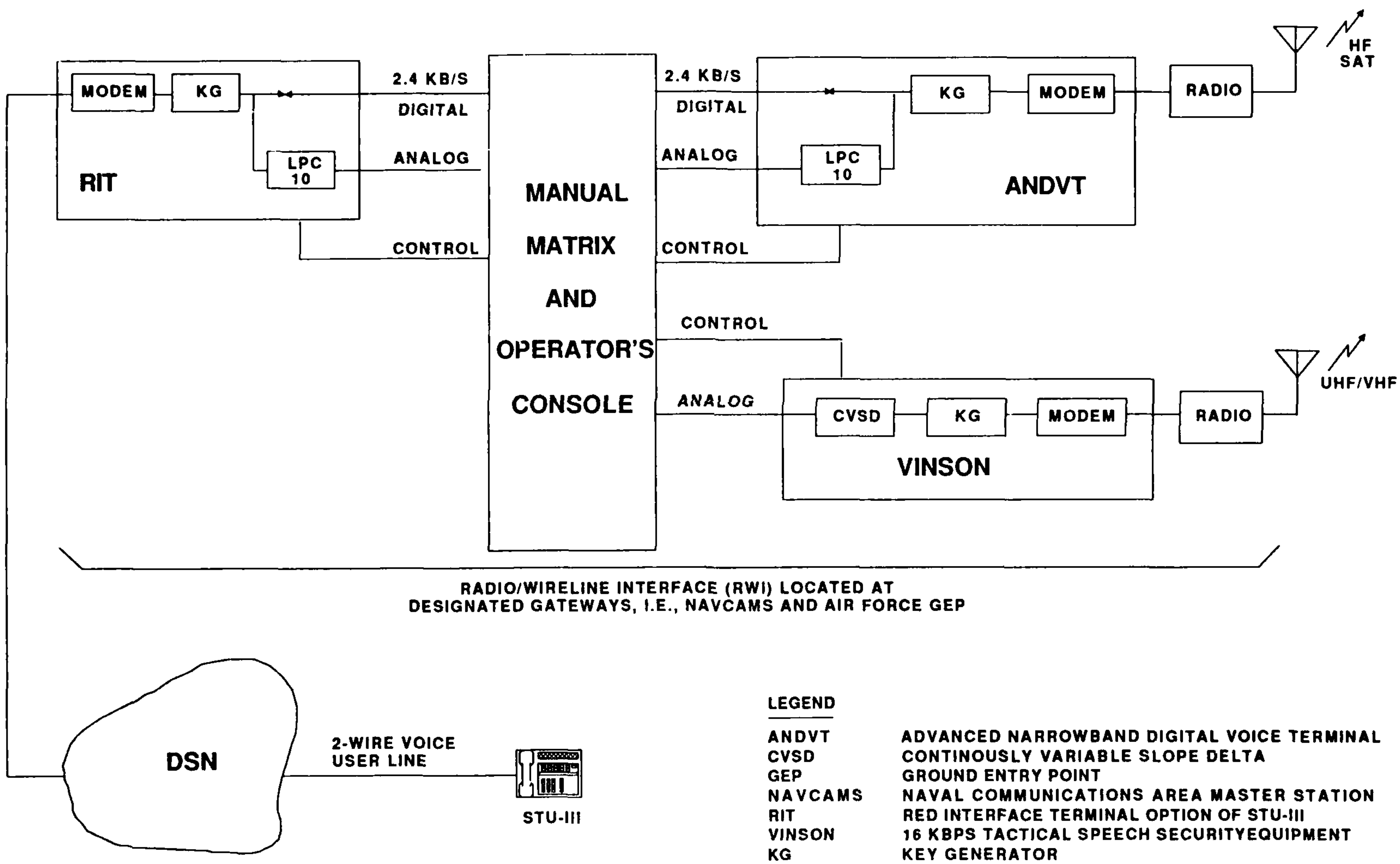


Figure 6. Tactical RWI Interoperability

RIT, as required, for operator initiation and receipt of STU-III calls and for the establishment and termination of interface connections. The console and its RIT's will be capable of interoperating with STU-III terminals in the secure mode. The console will provide for network push-to-talk control through digital speech detection for full-duplex connections, when required. For RED digital interfaces, the console will buffer timing differences between the wireline and radio links to minimize the probability of speech clipping. The console will provide for the automatic termination of a call upon receipt of a release message by the RIT or of a preempt signal.

3.6.2 RWI Call Description

Calls to network radio users are initiated by the STU-III user dialing the DSN number of the RWI. The call is routed through the network to an idle RIT at the RWI. The RWI operator answers and verbally obtains the necessary data (called party and caller capabilities) to complete the connection. The operator will then establish secure contact with the called party using normal radio network procedures and establish the interconnect. Once the connection is established, operator control will not be required.

3.7 STU-III INTERFACE TO TRI-TAC (AN/TTC-39)

Two interface options will be possible: the deployment of STU-III's with TRI-TAC users and the DSVT/STU-III interface capability at a RED Switch.

3.7.1 STU-III Deployment with TRI-TAC AN/TTC-39

The AN/TTC-39 switch has the capability to interface AUTOVON/DSN via analog interswitch trunks using standard AUTOVON/DSN signaling and supervision. Therefore, if STU-III's were terminated directly as subscribers on the AN/TTC-39 switch analog matrix, then calls could be completed between the DCS STU-III's and the tactical STU-III's in the same manner as calls between STU-III's in the DCS (see Figure 7). Tactical users would have to be equipped with STU-III's as well as their own Digital Subscriber Voice Terminal (DSVT). The capability of the STU-III to operate over the TRI-TAC system in this manner is being tested; preliminary results have been satisfactory.

3.7.2 Manual RED Switch Interface

The RED Switch can provide an operator-assisted interface capability between the DSVT subscribers served by a TRI-TAC equipped (AN/TTC-39) switched network and STU-III subscribers served by the DSN or commercial networks (see Figure 8). The RED Switch will interface the tactical network through a DSVT (CVSD algorithm) interface via access lines or trunks operating at 16 or 32 kb/s and will be capable of interoperating with any DSVT-compatible tactical secure voice element. The RED Switch will interface DSN through an RIT (STU-III) operating at 2.4 kb/s (LPC algorithm). The internetwork connection between the RIT and DSVT interface lines will be made by the operator at the RED analog level. This arrangement would require authorization by the RED Switch operating agency. The quality of voice, speed of service, and security procedures are affected.

3.7.2.1 Quality of Voice

Voice quality between a STU-III user and a DSVT user at 32 kb/s is expected to be fair to good. Connections with the DSVT at 16 kb/s would probably be poor to unacceptable.

3.7.2.2 Speed of Service

Since all calls will be manually processed, the call set-up time will be significantly extended.

3.7.2.3 Security

STU-III calls to and from the RED Switch will be automatically restricted to terminals that are keyed (CIK inserted) SECRET or higher. Calls extended to or from tactical

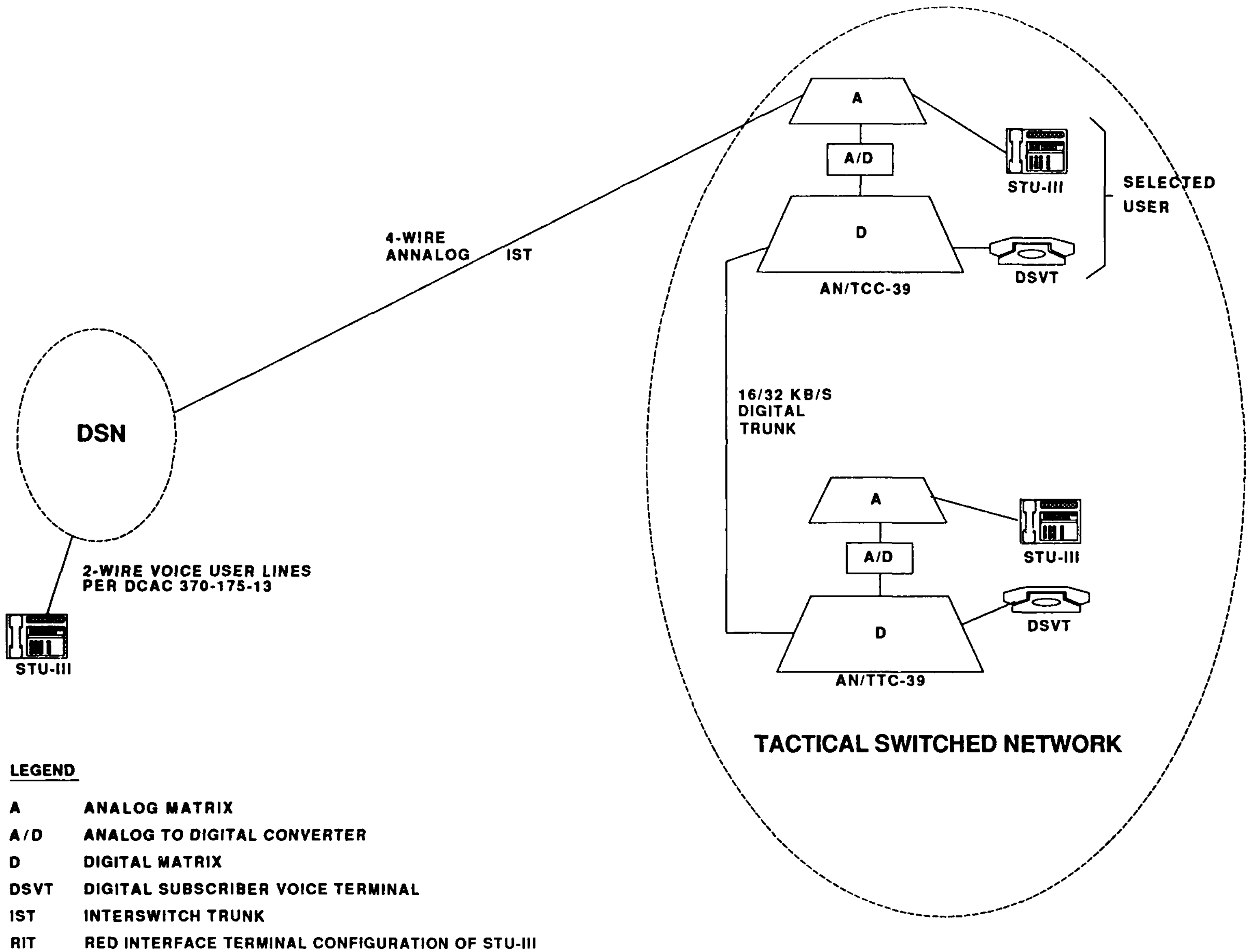
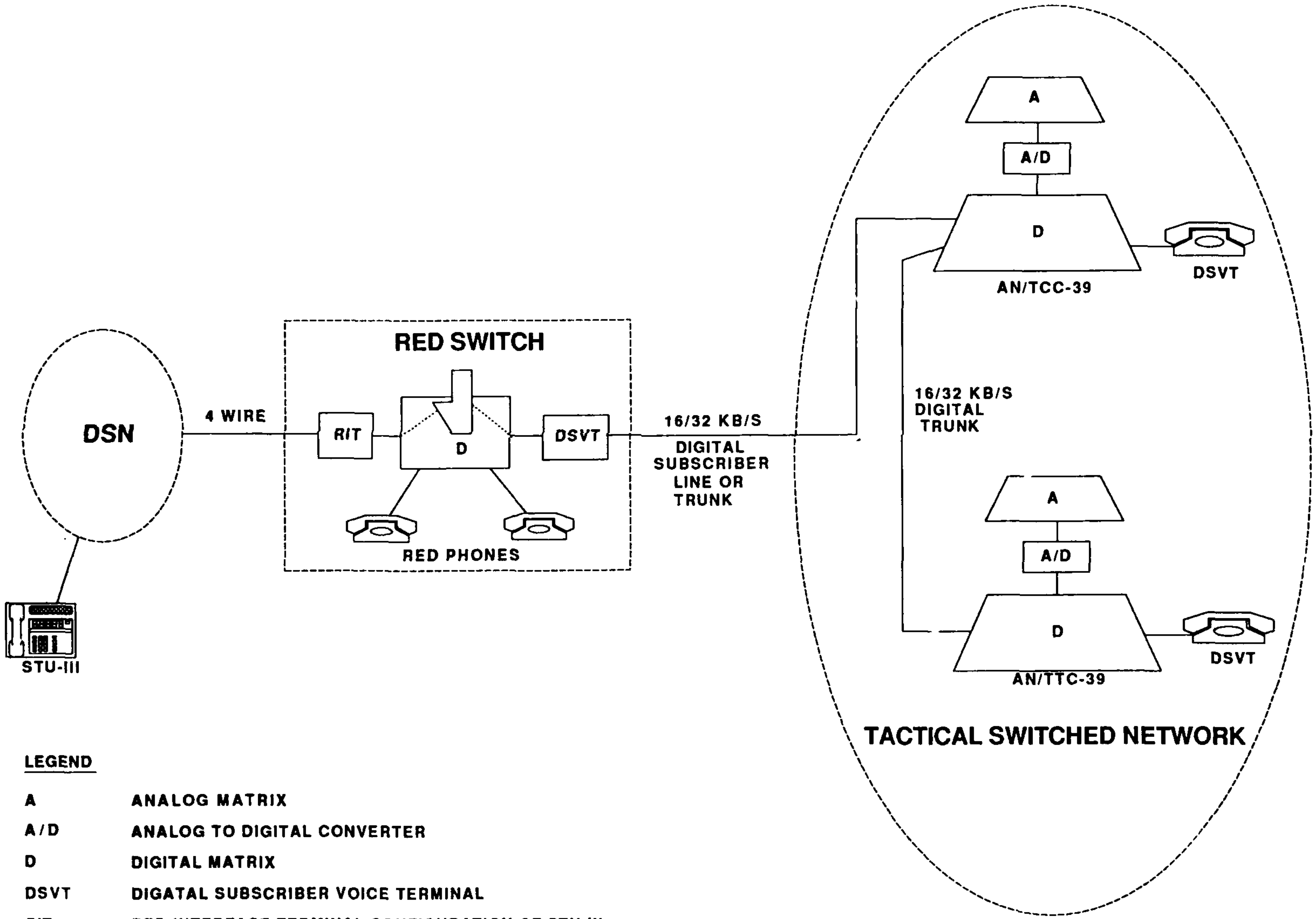


Figure 7. STU-III/TTC-39 Interoperability (1)



LEGEND

A	ANALOG MATRIX
A/D	ANALOG TO DIGITAL CONVERTER
D	DIGITAL MATRIX
DSVT	DIGATAL SUBSCRIBER VOICE TERMINAL
RIT	RED INTERFACE TERMINAL CONFIGURATION OF STU-III

Figure 8. STU-III/TTC-39 Interoperability (2)

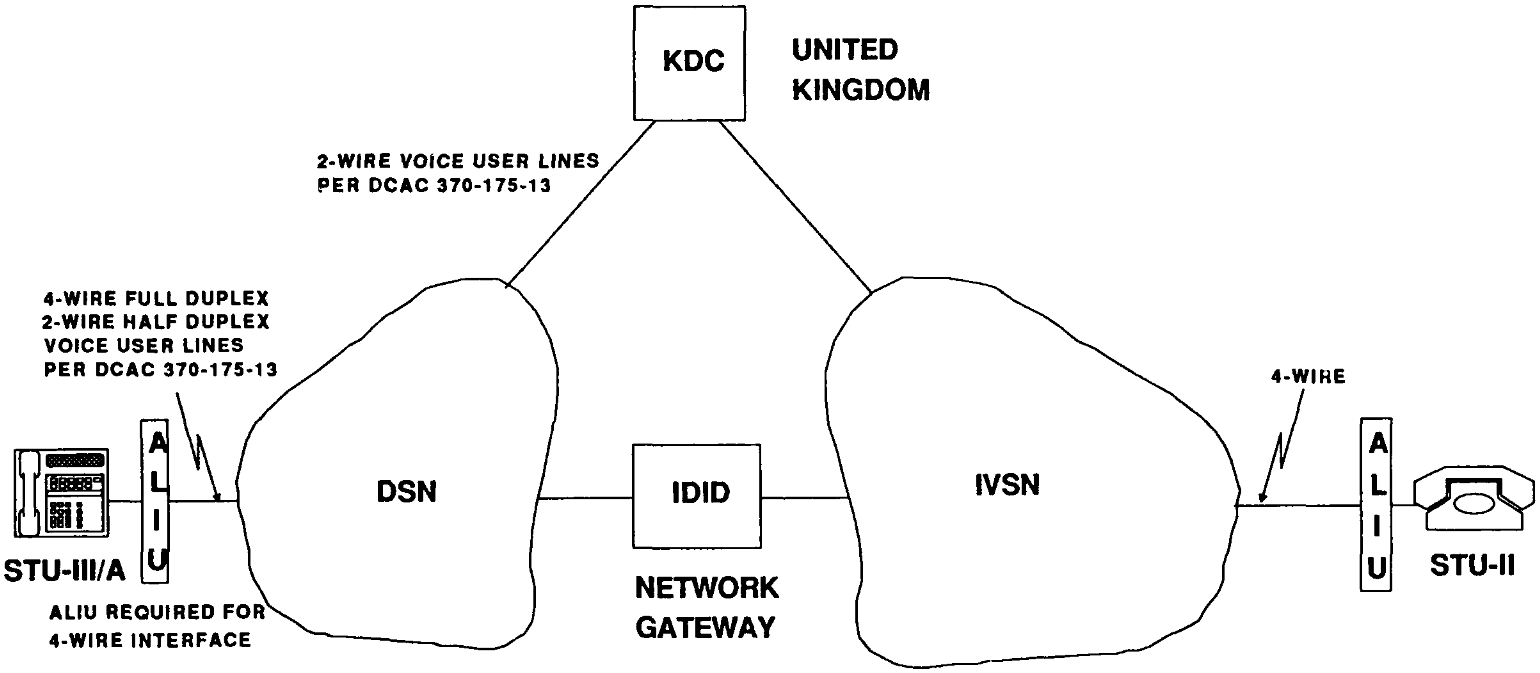
users will also have to be limited to terminals that are keyed SECRET or higher.

3.8 NATO INTERFACE

The planned NATO secure voice capability is based on the use of STU-II type terminals and BELLFIELD keying. NATO interoperability for a limited set of DoD users will be provided by the STU-III/A, which has a STU-II compatible voice algorithm at 2.4 kb/s and both the BELLFIELD and FIREFLY keying processes. The NATO STU-II will be terminated as a four-wire drop on the NATO Initial Voice Switched Network (IVSN) and will utilize a four-wire line adapter identical to the AUTOVON Line Interface Unit (ALIU) (see Figure 9). The STU-III/A must also be terminated as a four-wire drop on DSN, utilizing an ALIU, if full duplex operation is required.

The IVSN and AUTOVON/DSN will be interfaced through a device called the IVSN/DSN Interface Device (IDID). The IDID can be thought of as a subscriber on both the IVSN and AUTOVON/DSN networks. In this role, the IDID essentially answers a call from either network and then goes off-hook to the other network, allowing a call to be placed into the other network after receiving a second-dial tone. The interface will permit precedence calls to be placed through both networks from either IVSN or AUTOVON/DSN subscribers.

The calling process from either network follows: Each IDID gateway will appear as a subscriber terminal and will have a network address; each circuit will be classmarked at the highest precedence level authorized. In a call between a STU-III/A and a NATO STU-II terminal, the BELLFIELD two-call procedure will be used. This procedure requires that the calling user dial the called party's five-digit ID number, its precedence (if applicable), and then the number of the gateway. The calling terminal will then automatically dial the BELLFIELD Key



LEGEND

ALIU	AUTOVON LINE INTERFACE
IDID	IVSN/DSN INTERFACE DEVICE
IVSN	INITIAL VOICE SWITCHED NETWORK
KDC	KEY DISTRIBUTION CENTER

Figure 9. NATO Interoperability

Distribution Center (KDC) to obtain key information and establish a second call to the called gateway IDID. When the second dial tone is heard, the caller will enter the precedence digit, if applicable, and number of the subscriber in the destination network. Since the STU-III/A and STU-II are directly compatible in the BELLFIELD mode, they can then automatically synchronize and go into the secure mode. The U.S. STU-IIIA must be classmarked at the KDC to allow interconnection to the NATO STU-II.

The NATO STU-II's do not have the capability to enter a precedence digit after a second dial tone (unless the terminal is modified or a separate DTMF key pad is added). Therefore, the STU-II user will be able to use precedence only to get to the gateway, not into the interconnected network.

SECTION 4 - TRANSMISSION

The STU-III modem design incorporates both near- and far-end echo cancellation; thus, it can provide full duplex operation over two-wire access lines as well as four-wire lines. Conditioned lines are not required; however, the following network aspects should be considered:

1. Network echo-suppressors and echo-cancellers should be capable of being tone disabled. The STU-III sends a disabling tone during its initial set-up protocol.

2. Tandem links of ADPCM should not exceed four (CCITT Recommendations G.113 and G.721).

3. Tandem links of satellite transmission should not exceed two.

4. The effects of TASI on the STU-III are still being tested; however, the effect of the STU-III on TASI will be to reduce the efficiency of the TASI, since the STU-III line signal is a full time data signal in the secure mode.

SECTION 5 - STATUS OF STU-III LOGISTICS SUPPORT PLANNING

5.1 LOW COST TERMINAL (LCT)

Present planning for logistics support of the STU-III/LCT terminals indicates that, in most cases, the terminals will be contractor-maintained and equipment substitution will be utilized. Support packages differ from vendor to vendor. The STU-III/LCT procurement contract was signed on 3 July 1986. A one-to-two-year warranty (vendor dependent) will be provided to the users under this contract, with a five year warranty option. The procurement contract provides users with the options for initial line and power installation. See the GSA Interdepartmental Logistics Support Plan (IDLSP), 25 January 1988, for more details.

5.2 STU-III/A AND STU-III/MPT

The IDLSP for these terminals has not been published. The contract signed on 1 March 1988 requires that the equipment be logistically supportable and that the contractor's maintenance procedures support the equipment in its fixed plant domestic or foreign operational environment. A one-year warranty with an option for an extended five-year warranty is also provided.

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7. CCITT Red Publication for International Standards, October 1984.

GLOSSARY OF TERMS AND ABBREVIATIONS

TERM/ABBREVIATION	DEFINITION
ALIU	AUTOVON Line Interface Unit
ANDVT	Advanced Narrowband Digital Voice Terminal (2.4 kb/s), generally used for HF radio
ADPCM	Adaptive Differential Pulse Code Modulation (32 kb/s)
AUTOSEVOCOM	Automatic Secure Voice Communications
AUTOVON/DSN	Automatic Voice Network evolving to the Defense Switched Network
BLACK	A facility or compound that handles only unclassified traffic and/or encrypted classified traffic
C^2	Command and Control
CIK	Crypto Ignition Key
CPE	Customer Premise Equipment
DCS	Defense Communications System
DoD	Department of Defense
DSN	Defense Switched Network
DSVT	Digital Subscriber Voice Terminal for TRI-TAC
FSVS	Future Secure Voice System
IVSN	Interim Voice Switched Network (NATO)
KDC	Key Distribution Center (BELLFIELD)
kb/s	Kilobits per second
KMC	Key Management Center
KY-71	STU-II
LCT	Low Cost Terminal (STU-III)

GLOSSARY OF TERMS AND ABBREVIATIONS (Cont'd)

TERM/ABBREVIATION	DEFINITION
LPC-10e	Linear Predictive Coding with 10 spectral coefficients, enhanced
MLPP	Multilevel Precedence Preemption
PBX	Private Branch Exchange
PSN	Public Switched Network
PTT	Postal Telephone & Telegraph Network
PTT	Push-to-Talk
RED Phone	Standard unencrypted telephone certified to process classified voice traffic
RED Switch	Telephone Switch certified to process unencrypted classified voice traffic
RIT	RED Interface Terminal (configuration of STU-III for direct RED digital and analog interface applications)
RSP	RED Switch Project, DoD project for CINC RED Switches
RWI	Radio/Wireline Interface
STU-II	Secure Telephone Unit, Type II (KY-71)
STU-III	Secure Telephone Unit, Type III, Family (LCT, A, and MPT)
SVIP	Secure Voice Improvement Program
SVS	Secure Voice System
TASI	Time Assigned Speech Interpolation
TRI-TAC	TRI Service Tactical Communications
VINSON	KY-57/58 Combat Net Radio Voice Security Equipment generally for UHF radio applications
VOX	Voice Operated Switch for controlling half-duplex transmission

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